19-11-86

VARIOUS CIRCS

10 24 58
16 44
35 54 63 ERRORS
20

58 5AMP SURGE CONTROLLER

DECEMBER 86 - FEBRUARY 87 PRICE 85p

Electronics

THE MAPLIN MAGAZINE

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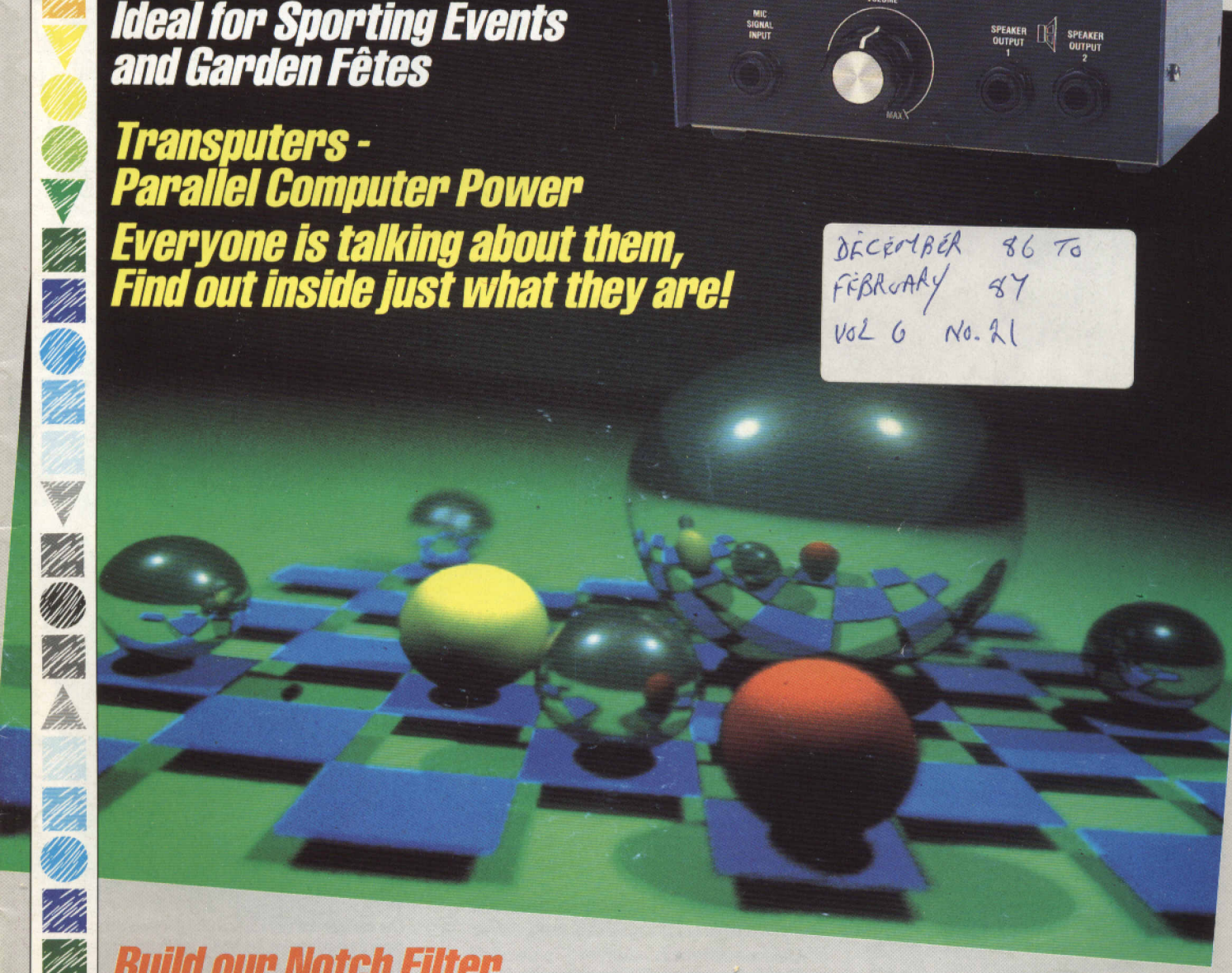
Transputers - Parallel Computer Power

Everyone is talking about them, Find out inside just what they are!

DECEMBER 86 TO FEBRUARY 87
VOL 6 NO. 21



Build our Notch Filter And Cut-out Unwanted Signals

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Watts in an Amplifier?

Which Resistor?

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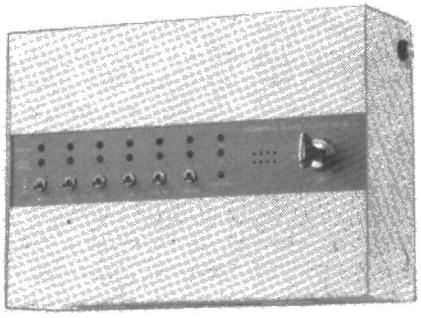

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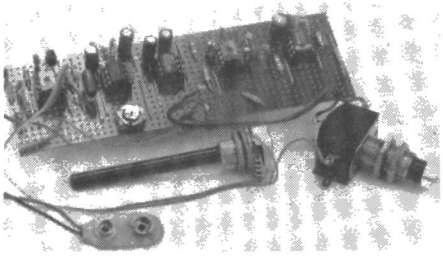

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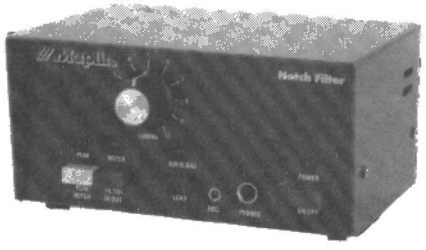

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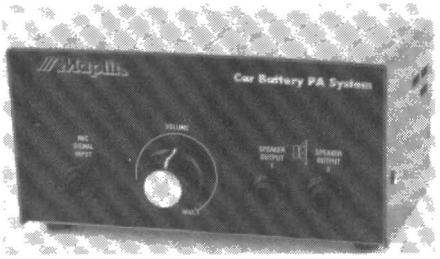

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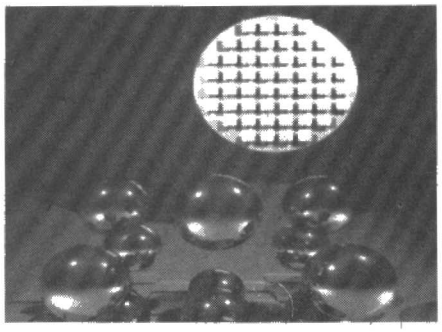

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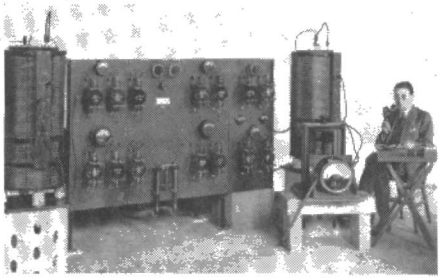

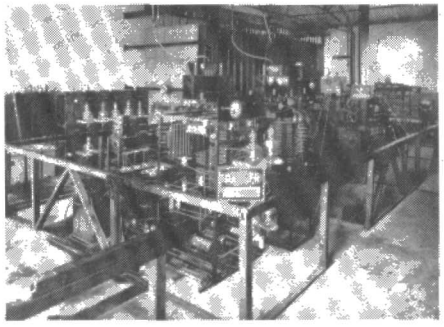

Part 5 of this series continues with the increasing development of the valve, and the subsequent radio boom of the 1920's.

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T·R·A·N·S·P·U·T·E·R·S

THE COMPONENT COMPUTER

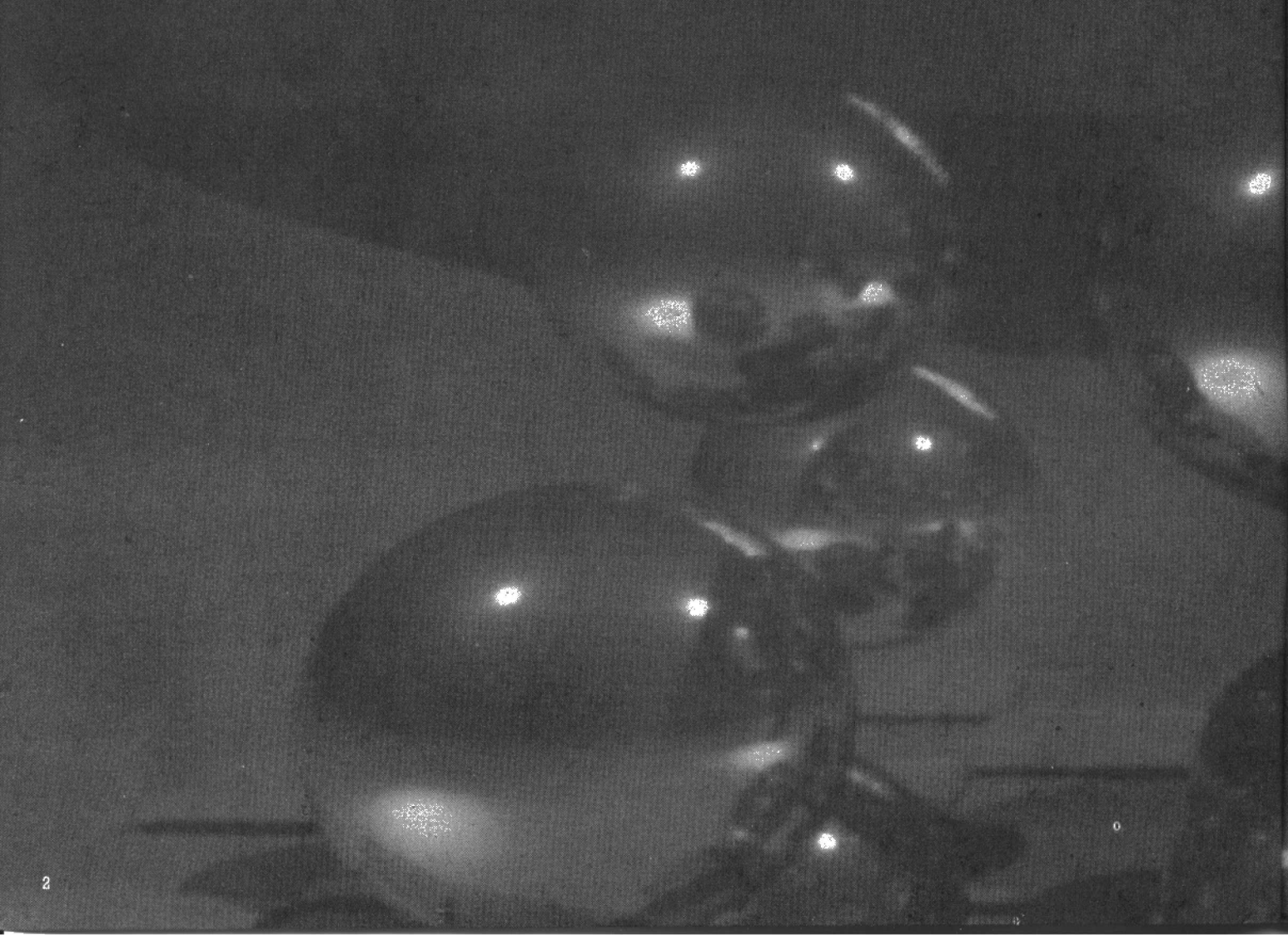
by J.M. Woodgate B.Sc. (Eng.), C.Eng., M.I.E.E., M.A.E.S., M.Inst. S.C.E.

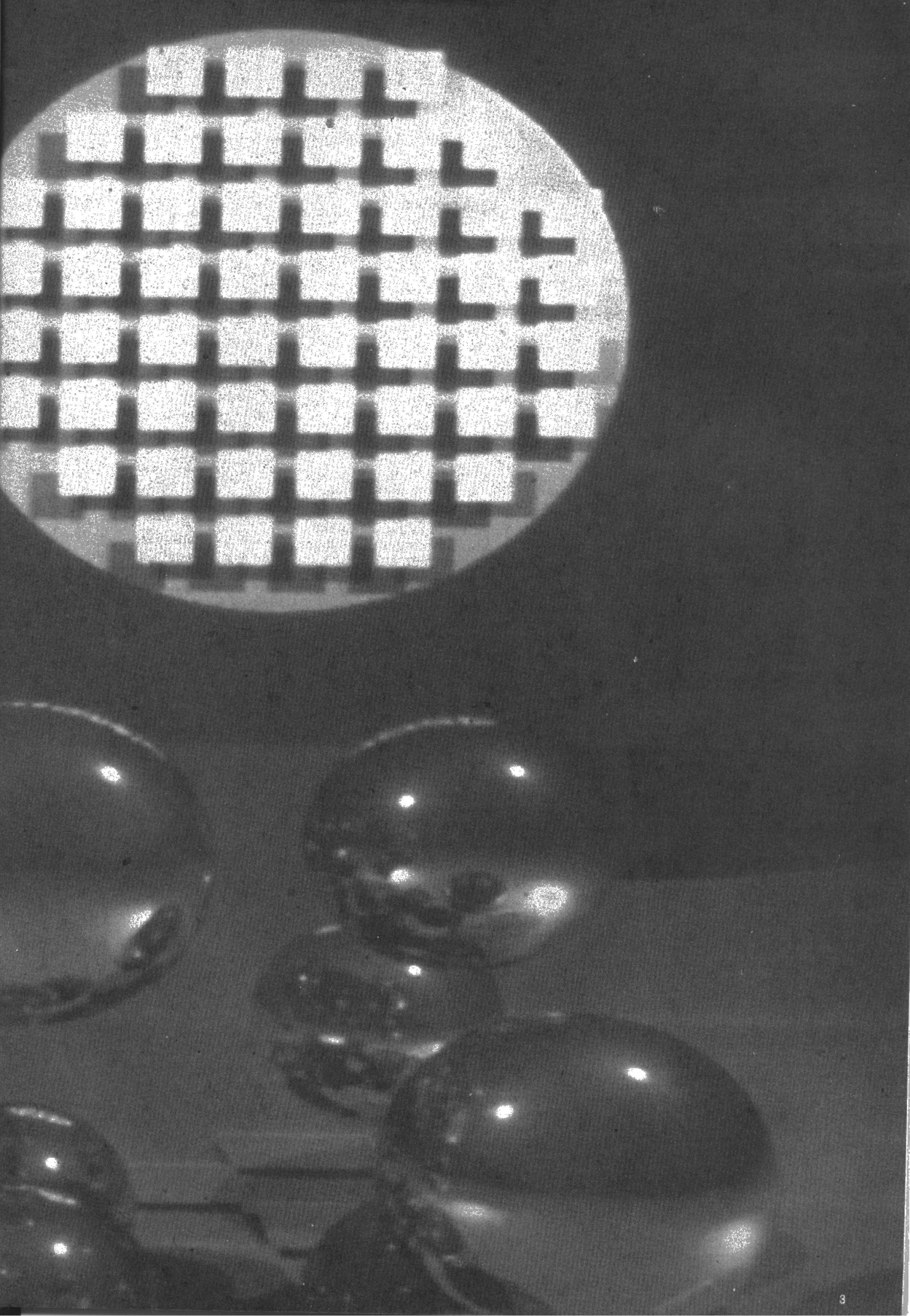
In the early days of electronic computing, even before the first real computer had been made, the possible structures of practicable computers were studied by John von Neumann, among others. It was he who established the structure, now known as the 'von Neumann architecture' which is common to virtually all digital computers designed so far. Figure 1 shows the elements of this structure.

In this article we shall examine what is possibly the first general-purpose digital computer device which breaks away from the von Neumann architecture, resulting in the possibility of very fast processing of huge amounts of data at a far lower cost. This device is the Transputer, and its understanding and efficient use requires also an appreciation of its specially-developed programming language, called 'occam', after the 13th-century philosopher, William of Occam. William was the first to state clearly the principle, known as 'Occam's Razor', that all scientific theories and mathematical processes (including computer languages) must be kept as simple as possible.

The von Neumann Bottleneck

When a von Neumann computer is working, its operations are controlled by the program counter, which forces the computer to follow through the program stored in memory. The program is a series of steps, each of which involves the transfer of a (binary) number from a specified place in memory to the central processor, where something is done to it, and in the next step the result is processed further or stored in memory, and/or sent to the output device. The central processor can actually do only very limited things to a number in one program step. It can store the number in memory, or onto a 'stack' (temporary store), add another number to it, compare it with another number (a form of subtraction) or multiply or divide it by two. What is most important is that while this is going on, all the other numbers in the memory, and all the other program steps, are simply sitting and waiting. In the early days, when programs were short and memories were small, holding only a thousand numbers or so, this could be accepted. But now that we have long





programs and memories of relatively enormous capacity (possibly well over 1 million numbers, even in a microcomputer), the use of a single central processor can be seen to cause a major bottleneck.

More Than One Processor

One way of overcoming this is to use more than one central processor, and this technique has worked quite well in small computers, such as the 'Wireless World' Scientific Computer and the Commodore 400. These use a special processor to do arithmetic: the same sort of processor as in a calculator. This is helpful, both in terms of speed and precision, because general-purpose processors, such as the Z80 and 6502 and their relatives, are naturally designed to perform a wide range of tasks using logic operations, rather than be particularly quick at arithmetic. Obviously, they can be programmed to do it, but are slow, especially if high accuracy is required. The 400 also uses a separate processor for driving the screen display: this technique is now common, but the devices used are usually called 'VDU controller', 'graphics controller', etc., rather than being regarded as an additional central processor.

This technique of using supplementary processors has certain limitations. On the one hand, most programs do not call for a great deal of arithmetic, so the cost and complication of the 'number cruncher' may not be justified, and on the other, the supplementary processor has to be controlled by, and feed numbers to, the main processor, thereby interrupting other work that the latter could be doing.

Parallel Processing

It may well be clear now where this line of reasoning is leading. Why not divide up the program into parts, each of which is handled, simultaneously, by a separate processor with its own chunk of memory, input and output devices? Well, until last year that wasn't economic: building such a parallel-processing computer with, say, Z80's was certainly possible and had been done in the course of research, but commercially it wasn't practicable. Even the defence industry has tended to balk at the cost of parallel processing computers using conventional processors, which is increased by the necessity of developing appropriate programming languages, based on the less-than ideal instruction sets of processors not designed with parallel-processing as a major requirement.

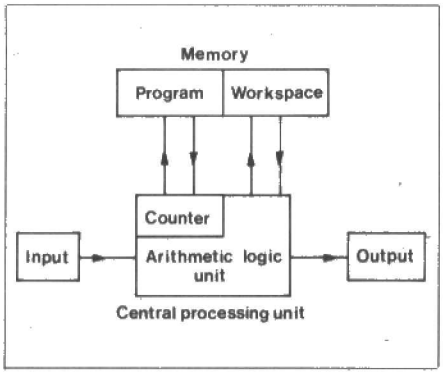


Figure 1. Block diagram of a von Nuemann computer.

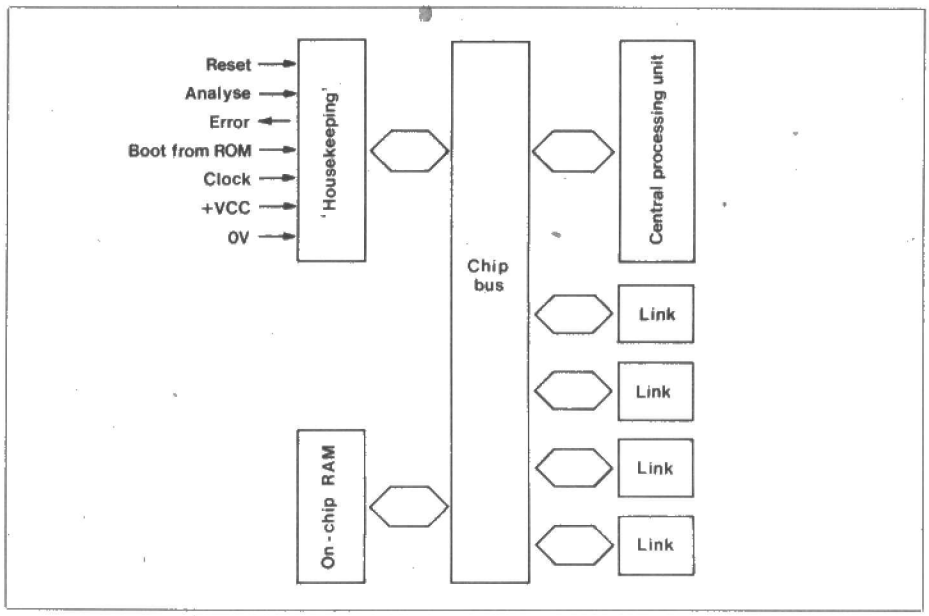


Figure 2. Block diagram of a Transputer.

Another way of improving the efficiency of a computer (i.e. being able to use a smaller computer to run a given program in a fixed time, or to run it faster on a given computer) is called 'multitasking'. This is useful, in a von Neumann computer, only if the program divides nicely into one or more main, or 'foreground' tasks, and a series of relatively quickly-executed background tasks.

The Transputer

In 1985, research carried out by Inmos, in this country and the USA, produced the Transputer. All on one silicon chip (see Figure 2) are a central processor, some memory and four communications links. Each of these links is both an input and an output device, designed to be directly connected to another transputer, or through a 'link-adaptor' to another type of processor or a special-purpose transputer, configured perhaps as a graphics controller or a disk controller.

Curiously, the closest approach to this type of device is possibly the central processor device in the ITT digital TV chip-set, discussed in a previous issue ('The Digits are Coming!', Maplin Magazine Volume 4 Issue 15). This also has a processor, some memory and communications links on one chip, but is heavily hard-wired internally for its specific functions in a TV set. At least, that is ITT's intention. Perhaps some ingenious experimenter may find unexpected flexibility.....?

The 'Process'

Much of the information published about the transputer is written in language which is closely oriented to those in the computer industry, and is somewhat impenetrable otherwise. Differences in authorship also complicate matters: the same concept may be expressed in totally different ways. However, each new technology generates new technical terms, and the transputer is no exception. The first new term is the 'process'. This is both an independent computation, with its own program (or part of a program) and data, and also the transputer that does that computation. It is difficult, at first, to see how the one word can mean both, but the idea does work, so we just have to accept it at this stage. A collection of processes is itself a process, and the complete set of all the processes is the whole program.

Concurrency

The second technical term is 'concurrency'. This is not a forged £20 note, but a special case of parallel processing (itself a technical term) where the result of one process is used by another, and the result of that passed back to the first process or on to a third, and so on.

Networks of Transputers

A program that contains concurrent processes can be run either on a network of transputers, which pass their results to each other through the communications links, or on one transputer, using multitasking. It is thus possible to develop a program, intended to run very fast on a large network of transputers, on a development machine using fewer, or even just one, transputer.

Transputers can be connected into networks in different ways. Most

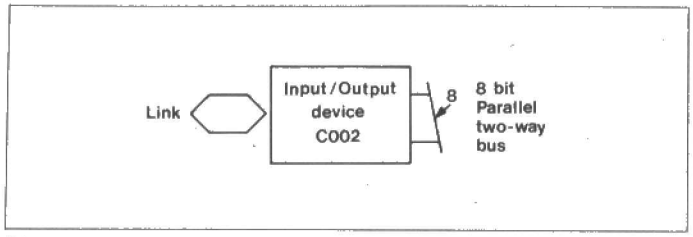


Figure 3a. Transputer I/O device.

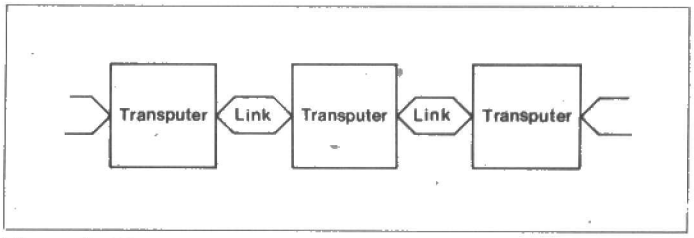


Figure 3b. Linear array.

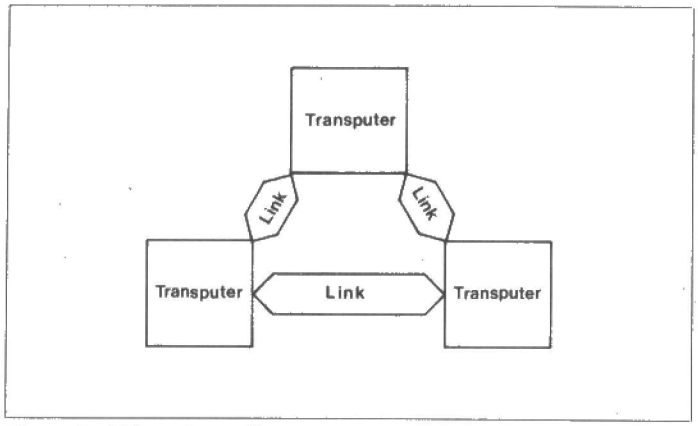


Figure 3c. Triangular cell.

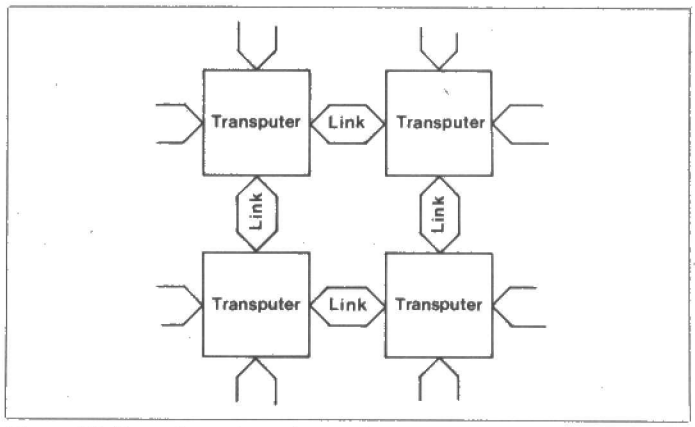


Figure 3d. Two-dimensional (square) array.

general-purpose transputers will have four two-way communications links, while special-purpose devices may have one or two. Devices with one link are input and/or output devices, and naturally come at the end of a line of devices, see Figure 3a. With two links a linear network of unlimited length is possible, see Figure 3b, or a single cell of a 2-dimensional triangular array, shown in Figure 3c, while four links allow an unlimited two-dimensional network, or square array (Figure 3d), an unlimited three-dimensional network (Figure 3e), with the same structure as a diamond crystal, and even a single cell of a four-dimensional array, shown in double perspective in Figure 3f! In this last example, each of the five transputers can communicate directly with each of the others, but it is necessary to cheat a bit in order to provide input and output links, although there is no reason why transputers should not be made with more than four links.

The Communications Link

Data for transmission through a link to another device can be obtained from the source transputer by direct memory-access (DMA), this being the fastest method overall. Data transfer can thus take place without interrupting the processor.

Links run door-to-door from one transputer to another, and thus resemble a car rather than a bus. They are two-way, so there is never any contention between devices, whereas in a bus system two devices are often trying to drive the bus simultaneously. Also, there are no bus loading problems: each link normally has one receiver only. Effectively, then, the standard data rate (10Mbit/s) of one link is multiplied in proportion to the number of transputers in the network, so the system can never run out of communication capacity. Furthermore, there need never be any transit-time problem: all data transfers can be over physically short paths, particularly if higher-dimensional networks are used.

Each link is two-way and each path is called a 'channel' in occam. Data is sent in serial form, with link control bits added, a start sequence '11' and an end sequence '0', see Figure 4. As soon as the receiver detects '11' on the input channel, and is ready to receive data, it sends an acknowledge signal, '10' on the associated output channel. This system automatically ensures reliable, synchronous data transfer, independent of word length and clock phase. It might appear that all transputers that are linked in this way would have to use the same clock, but in fact there is a frequency tolerance of 400Hz between any two separate 5MHz clocks: this allows the use of low-cost 200ppm crystal clock oscillators. Since the links are serial, they consist of only two conductors, one for each channel, and this makes the layout of printed

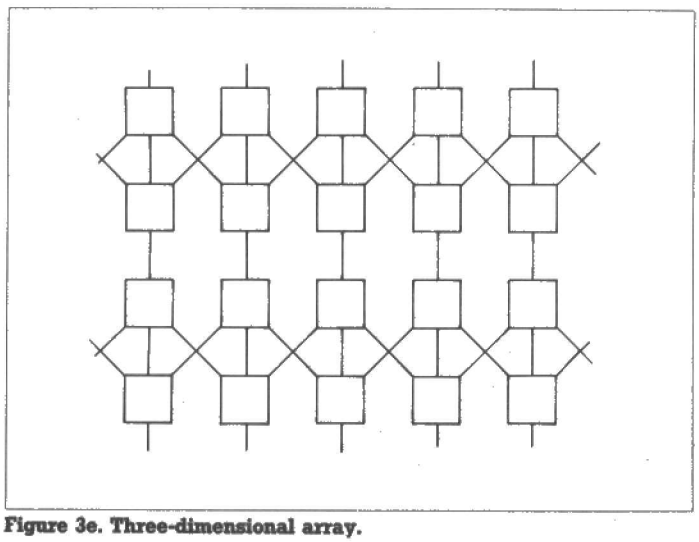


Figure 3e. Three-dimensional array.

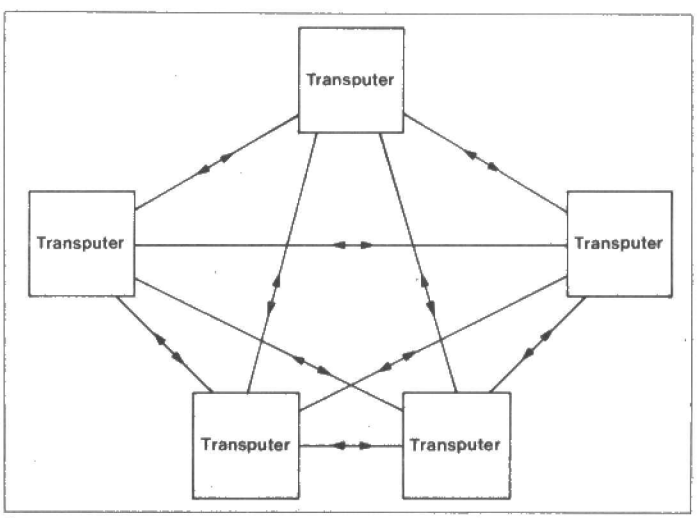


Figure 3f. Four-dimensional (pentatope) array.

Transputer created image.

circuit boards simpler. Imagine trying to lay out the network of Figure 3f for a 16-bit parallel bus!

The 10MHz link data rate and the 5MHz clock frequency are common to all transputers. The intention is that when faster devices are introduced, they will still communicate with earlier types, although they may talk to each other faster. All transputers generate fast 4-phase on-chip clocks from the standard clock input.

On-chip and Off-chip Memory

For some purposes, the transputer may need only its on-chip RAM. With current devices, 2k byte of RAM is provided, with a parallel bus interface and a maximum data rate of 20Mbit/s on each bus line. However, general purpose transputers also have another parallel interface, for their own off-chip memory, up to the maximum permitted by the processor, i.e. 64k for the 16-bit T212, and 4G (4 294 967 295) for the 32-bit T414.

Naturally, the data rate associated with off-chip memory is rather lower than for the on-chip variety, but is still around 5Mbit/s per bus line. In programming terms, the two sorts of memory are identical, but there may in extreme cases be a speed penalty of 2 to 3 times if repeatedly-used data, or a highly repetitive program, is held off-chip.

The 16-bit T212 external-memory interface has a fixed configuation, with a 16-bit address bus and a 16-bit data bus. External devices may request control of

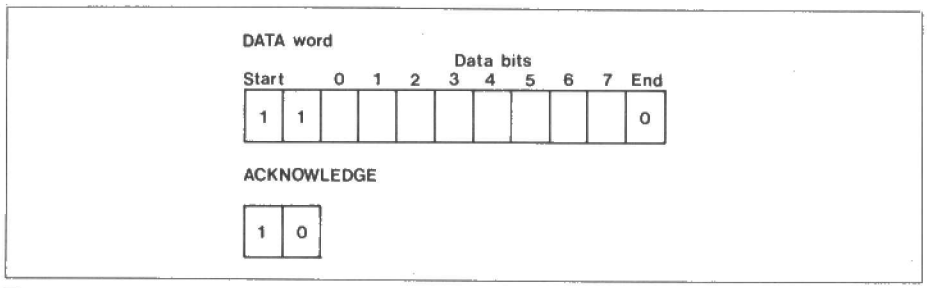


Figure 4. Link byte formats.

the whole interface for DMA transfer, while the processor continues to operate using on-chip memory. The 32-bit T414 interface is much more versatile. Address and data are multiplexed, and, on start-up or reset, the interface can be configured in 13 different ways. The configuration can be hard-wired at the chip socket, or derived as a 36-bit word from external memory. Timing and refresh signals for multiplexed dynamic RAM are all provided, if required, by the transputer, so that no extra devices are needed.

Timers

Present transputers include at least two timers. These are essential for full use of the occam language, which includes the concept of levels of priority for different processes.

Arithmetic

Unlike the general-purpose single processors mentioned before, current transputers can cope well with arithmetic, and support floating-point operations at 100kFLOPS (100 thousand Floating-Point Operations Per Second). Devices soon to be released will have hardware-support for floating-point, and thus will not depend on repeated software access, so they will be up to ten times faster.

Error-handling is quite comprehensive, with automatic checks on array bounds, arithmetic overflow detection, etc. It is possible to 'freeze' most of the data in the transputer when an error occurs, so that the cause of the error can be investigated and, with luck, determined.

Logical Behaviour

The occam language has been designed in such a way that every aspect of the program which is not affected by real-time effects is independent of the way in which tasks are allocated to the transputers in an array, and independent of processing and communication times. This is, of course, essential if large programs are to be developed on small arrays.

Programming the Transputer

Inmos decided that the transputer should be programmed in high-level language, i.e. a language that people can understand fairly easily. With most processors, high-level languages have limitations and the really exciting things have to be written in machine-code, which is a good deal more difficult to decipher, something like Japanese kana characters in complexity. A 'half-way house' is provided in the form of 'assembler' language, which is easier to understand but no less long-winded to write.

The transputer machine-code uses a minimum number of 'instructions'. An 'instruction' is more or less what the work normally means; it tells the processor to do something. Having a small number of instructions makes machine code programs very long, but makes the translation of high-level language into machine code more reliable. More on this later.

Instructions

On the present transputers, instructions are 8-bit words, divided into 4 'function' bits and 4 'data' bits. The four 'function' bits can represent sixteen functions, as shown in Figure 5. We can call these 'simple' functions, and thirteen of them are the fundamental operations of computing. The remaining three have special functions. 'Negative prefix' simply causes the data to be treated as a negative number. 'Prefix' allows data words of up to 32 bits to be built up from the 4-bit data 'nibbles' of successive instructions. 'Operate' causes all 8 bits of the instruction to be treated as a function code. Clearly, this can be a new function, extending the function set up to 255 members. By using both 'Prefix' and 'Operate', 32-bit instructions can be built up, extending the set to over 4 billion members! This allows ample room for any foreseeable future development, and makes it possible to write programs now, which can be guaranteed to run on processors which have not yet been developed!

Coping with High-level Languages

There are two ways that a computer can use to translate a program written in high-level language into machine-code that it can understand and act on. In the first method, each program line is looked at as it comes to be used, and is translated into machine code at that time. This is called 'interpretation', and it is a very flexible and powerful method (e.g, BASIC). But the translation operates in real time, i.e., the running time of the program includes the translation time. This is most significant if the program is highly repetitive; in fact a commonly-used way of temporarily holding up a program is to include an 'empty' loop

★ Load Constant
★ Add Constant
★ Load Local
★ Store Local
★ Load Local Pointer
★ Load Non-Local
★ Store Non-Local
★ Jump
★ Conditional Jump
★ Call
★ Prefix
★ Negative Prefix
★ Operate
★ Reserve
★ Reserve
★ Reserve

Figure 5. The sixteen 'simple' functions.

which is traced thousands of times. Wouldn't it be better to do the translation into machine code once only, in 'spare time', and store the machine code to use in future instead of the original program? Well, this can be done, and it is called 'compiling'. The snag is that compiling some high-level language operations is difficult, especially if the processor has a large instruction set, and the compiling process may produce machine code that will run, but is much longer than necessary, so that the full speed improvement cannot be obtained.

The Occam Compiler

Since it was always the intention that the transputer would not be programmed directly in machine code, but in a high-level language, preferably occam, the 'simple' instruction set and the occam language itself have been designed so that the compiling of occam is both flexible and efficient. It is also possible to compile nearly all of the well-used high-level languages efficiently into transputer machine-code, and compilers for Fortran, Pascal and C are available. Concurrent programs can be written in occam, with sub-processes in other languages.

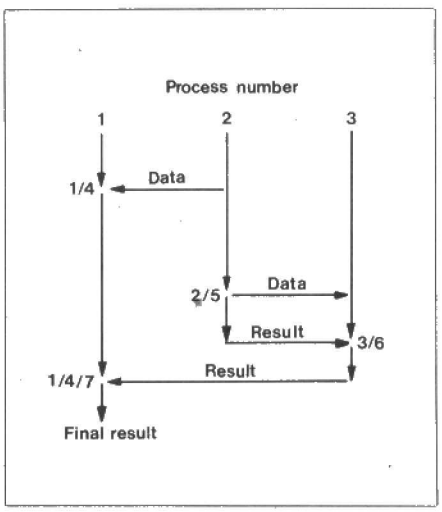


Figure 6. Concurrent processing executed serially.

Occam

Remember that the philosophy of occam is 'keep it simple'. There are only three fundamental processes in occam, out of which all others can be built. These are:-

INPUT.	Symbol '?' chan1?X means 'put X equal to the value coming in on channel 1'.
OUTPUT.	Symbol '!' chan2!Y means 'send out the value Y on channel 2'.
ASSIGN.	Symbol ':=' Y:=X*X means 'let Y = X-squared'.

We also need to tell the computer whether the process we are writing is concurrent or sequential. Simple processes are sequential, and are labelled as such:-

```
VAR X:
SEQ
  chan1?X
  chan2!X*X
```

This will take the incoming value on channel 1, square it and send it out on channel 2. 'SEQ' is called a 'constructor', and anything that follows it, up to the next constructor, is a single process. 'VAR X:' just 'declares' the variable X. This 'declaration' step is a requirement of many high-level languages, but not BASIC.

Occam uses 'WHILE' to form loops. There is endless argument over the 'best' way of forming loops, and there is no single answer. Providing the chosen method is quick and bug-free, it doesn't matter much in practice.

'PAR' is the constructor that introduces concurrency:-

```
PAR
VAR X:
chan1?X
  WHILE X<1000
    VAR Y:
    SEQ
      chan1?Y
      chan2!Y*Y
WHILE X>Y
  VAR Z:
    chan3?Z
    chan4!Z*Z
```

This will simultaneously calculate Y^2 and Z^2, subject to the conditions given in the WHILE expressions.

We can make the above fragment into a 'process', like a 'procedure' in BBC-BASIC, if we want to use it several times in a larger program:-

```
PROCtwosquares(CHAN chan1,chan2,chan3,chan4)
  PAR
    VAR X:
....etc.
```

The process is then called by using its name, 'twosquares', and the four channel parameters.

Instead of simultaneous calculations, a concurrent program may be required to branch, depending on which of several possible input channels sends data first. This is arranged with the constructor 'ALT'. The presence of data

on a channel, regardless of its value, is detected by assigning the input to the dummy variable or identifier, 'ANY'. The declaration 'DEF' assigns constant values to variables.

Naturally, there is a good deal more to occam than the above, but the fundamentals have been covered, and a good introductory programming manual can be obtained from Inmos.

Using the Transputer

A useful insight into the merits of parallel processing can be gained by considering how a concurrent program could be run on just one transputer. Suppose that the program divides nicely into three parts, which have to exchange intermediate data (e.g. part 1 needs a figure calculated by part 2 on the way to its final result). When the program is run, part 1 is executed up to the point where it needs data from part 2, and is then temporarily stopped. This condition is called 'sleep mode'. Part 2 then runs until data required by part 3 has been calculated. Part 3, however, is not ready to receive the data, as it has not yet run. So part 2 goes into sleep mode, and part 3 runs up to the point of receiving the data from part 2. Part 2 could now restart, but part 1 has been waiting longer, so, unless part 2 has priority, part 1 now runs until it requires data from part 3, when it goes into sleep mode. Part 2 can now run to its end, passing its result to part 3, when it has run to the appropriate stage. Part 3 then runs to its end, passing its result to part 1, which can then run to produce the final result. The diagram in Figure 6 helps to show this process.

If this program were run on an array of 3 transputers, all three parts could begin at the same time, and data exchange would occur between the parts with very little waiting time, perhaps none.

Practical applications of transputer arrays are clearly those tasks which naturally require parallel processing. One such is 'finite element analysis', which is a numerical method of calculating fields, equally applicable to magnetic fields, winds for weather forecasting, heat flow for cooling nuclear reactors, and the sound radiation of loudspeakers. Perhaps the most familiar of this type of calculation is the 'update' process on a spread-sheet. Here, for example, there might be a hundred entries of VAT amounts, calculated from a single cell entry of the VAT percentage rate (15%). Suppose the VAT rate changes to 12% (hint to Chancellor!): you change the figure in the 'rate' cell to '12' and press 'U', or whatever, to update the calculations. This asks the computer to do 100 multiplications, which a conventional computer would do one after the other. But a 100-transputer array would do them all simultaneously, effectively working 100 times as fast as the ordinary computer. For the highest possible

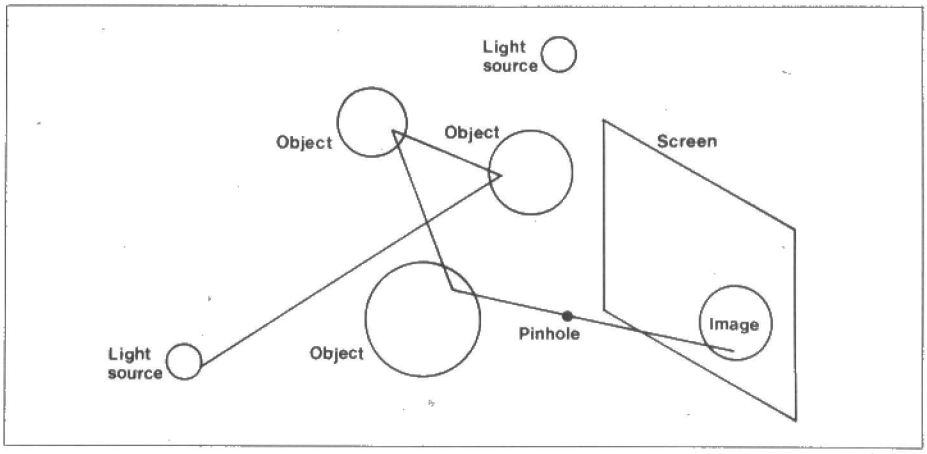


Figure 7. Principle of ray-tracing.

speed, each cell could have its own transputer, and the network of transputers would then be a 'map' of the spread-sheet itself. We can begin to see the close correspondence between the software process and the hardware process, i.e. the transputer that executes the software process.

Another application which is ideal for the transputer is computer graphics. Particularly if high resolution movement and perspective effects are to be produced, the number of calculations required per second can be astronomical, beyond the range of all but the fastest mainframe machines - or an array of 40 or so transputers, fitting into quite a small box and costing perhaps 1000 times less than the mainframe! One of these techniques is called 'ray-tracing', and an example of the results is shown in the front cover picture. In spite of appearances, the objects in the picture do not exist, nor is it a question of trick photography. The whole picture has been generated by an array of 40 transputers, and a program including the shapes and surface textures of the objects, the positions of the light sources and the laws of reflection. The calculation of the colour and brightness of each point of the picture is based on tracing a ray of light backwards from a screen, representing the plane of the picture, through a pinhole and back until it hits an object in the 'computer world', where it may be reflected or absorbed, or changed in colour, see Figure 7. Event-

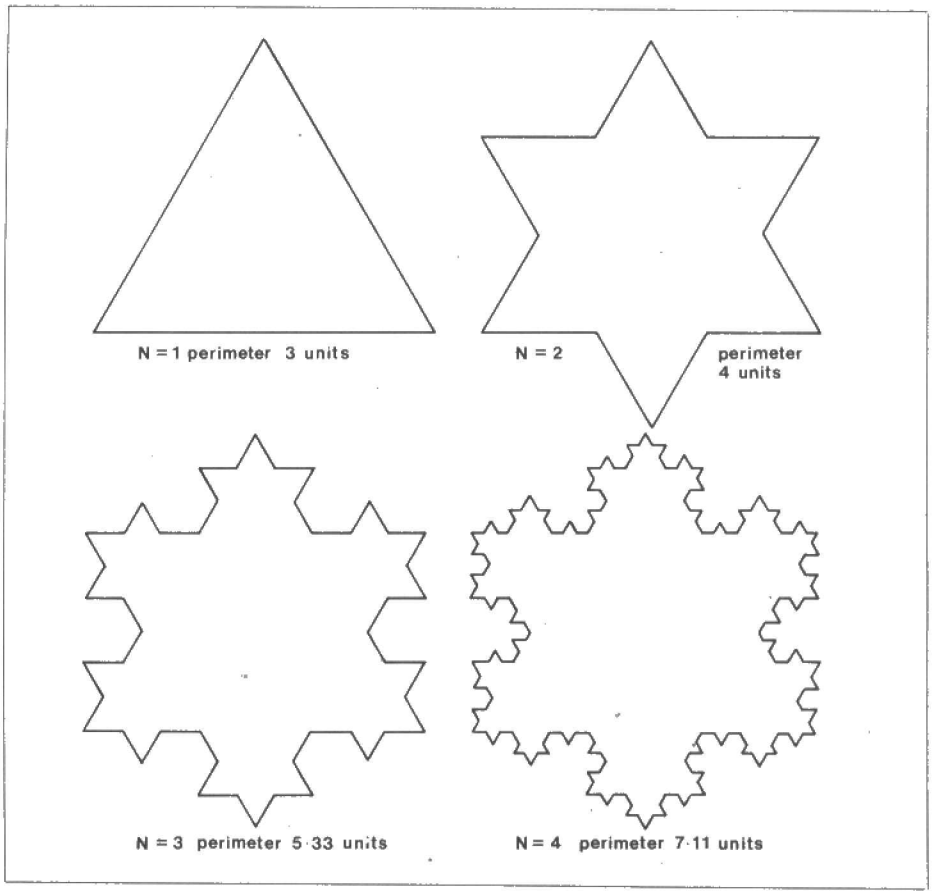


Figure 8. The first 4 iterations of the Koch snowflake. Each iteration increases the perimeter by a factor of 4/3. The 25th iteration has a perimeter nearly 100 times that of the original triangle.

ually the ray must end on a light source, or a black area of an object. Sixteen rays have been traced for each of 512 x 512 pixels, and hundreds of calculations are needed for each ray. Each ray can be calculated simultaneously on the transputer array, whereas they would have to be calculated one after the other on an ordinary computer.

Even more startling than the surrealist art which can be produced with these techniques are the naturalistic effects produced by what is known as 'fractal geometry'. Figure 8 shows a very simple example of this, known as the 'Koch snowflake'. We start with an equilateral triangle, and replace the middle part of each side by a triangle of one-third the size, rubbing out the internal lines. This process is then repeated indefinitely, producing a closed curve which, however much it is magnified, still has the same fine structure, and is infinitely long. Coastlines, especially those like that of Norway (thanks to Slartibartfast), are natural examples of fractal geometry. By introducing a random element into a program for calculating fractal surfaces in 3 dimensions, extremely realistic mountain and coastline scenes can be generated. Another, related, mathematical concept is the Mandelbrot set, discovered by Benoit Mandelbrot while investigating what are known as 'Julia fractals'. These represent the boundaries of sets of points $z_o = (x_o + jy_o)$ in a plane, chosen so that when the process $z_n = z_{n-1} + w$ is repeated indefinitely, z_n remains finite. For the Mandelbrot set, $w = z_o$. These sets would have remained a mathematical curiosity, in spite of the fact that the boundaries, being fractals, are infinitely complex, if parallel processing had not been developed. As it is, the generation of displays of the Mandelbrot and other related sets is used for estimating the efficiency of parallel-processing machines, and has opened a new field of computer graphics.

When Can You Have One?

The development of a device like the transputer is extremely complex and time-consuming. The cost of this development has to be recovered in the early stages of commercial exploitation of the product by pricing it as high as the market will bear. This has almost no relation to the material cost, of course; the material cost of a transputer is comparable with that of a 2N3055! But against this is the fact that microchip production is essentially a large-scale process, producing per week some hundreds to some tens of thousands of devices. To sell all these devices, the price must fall rapidly. So it is, and will be, with the transputer. Initial samples were released in September 1985, so the early sample phase is now over, and the T212 (16 bit) and T414 (32 bit) are now available on a normal commercial basis, at one-off prices of £271 and £338 respectively. A B004 plug-in for your IBM PC will set you back £2,900, but will add 1 or 2M byte of memory and, with the D700 software development system, allow programs to be written in occam and run on the PC. Also recently introduced is a product called K-MAX from Kuma Computers which consists of an add-on box attached to the 'ROM' port of the Atari ST and a dual processor board which contains an Inmos T414 transputer (can be expanded), and two blocks of 256 K-bytes of RAM, and is priced at £1450.00 plus VAT.

These prices are out of the range of most Maplin customers, but in a year from now the T414 could be £50 and the T212 as little as £30! Clearly, transputer projects will be quite feasible at that stage. Start learning occam now!

Acknowledgement

Thanks are due to Janice Lyons of Inmos for valuable information and pictures.

TOP TWENTY BOOKS

1. (1) Loudspeaker Enclosure Design and Construction. (WM82D) Cat. P60.
2. (4) Power Supply Projects, by R.A. Penfold. (XW52G) Cat. P56.
3. (3) Remote Control Projects, by Owen Bishop. (XW39N) Cat. P57.
4. (2) International Transistor Equivalents Guide, by Adrian Michaels. (WG30H) Cat. P52.
5. (7) Radio Control for Beginners, by F.G. Rayer. (XW66W) Cat. P57.
6. (10) IC555 Projects, by E.A. Parr. (LY04E) Cat. P58.
7. (6) How to Design and Make Your Own PCB's, by R.A. Penfold. (WK63T) Cat. P55.
8. (18) Electronic Security Devices, by R.A. Penfold. (RL43W) Cat. P57.
9. (8) How to Use Op-amps, by E.A. Parr. (WA29G) Cat. P54.
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11. (–) Practical Electronic Calculations & Formulæ, by F.A. Wilson. (RQ23A) Cat. P53.
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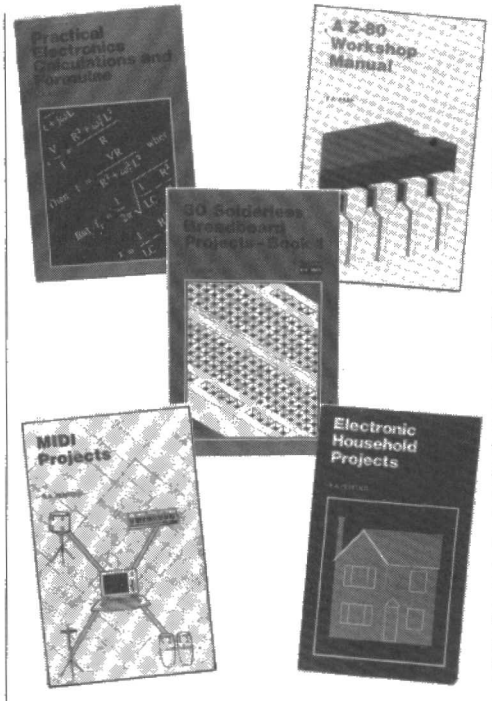
13. (11) Mastering Electronics, by John Watson. (WM60Q) Cat. P55.


Practical Electronics Calculations and Formulae
A Z-80 Workshop Manual
30 Solderless Breadboard Projects - Book 1
MIDI Projects
Electronic Household Projects

14. (16) Oscilloscopes: How to use them, how they work, by Ian Hickman. (WG34M) Cat. P58.
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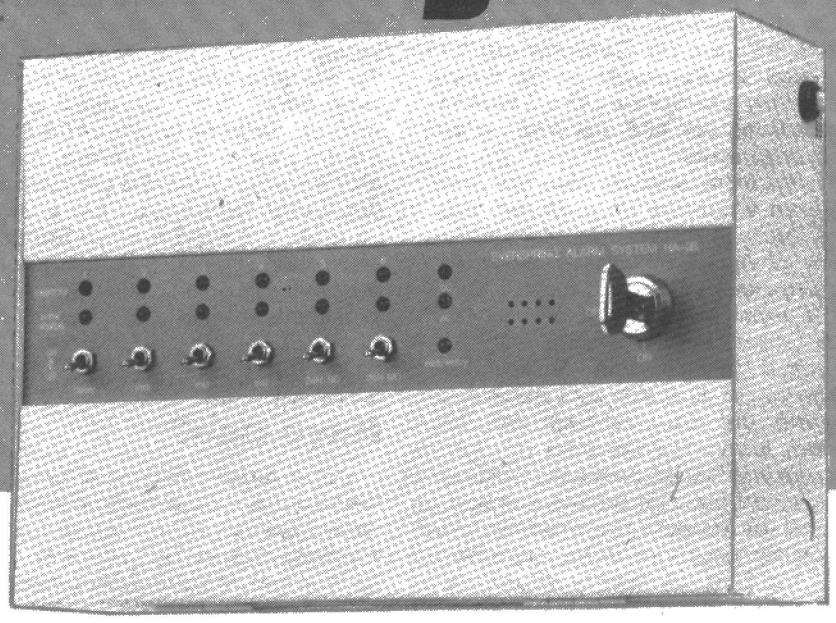
18. (17) Audio Projects, by F.G. Rayer. (WG46A) Cat. P59.
19. (–) Electronic Household Projects, by R. A. Penfold. (XW44X) Cat. P56.
20. (15) 50 Simple LED Circuits Book 2, by R.N. Soar. (WG43W) Cat. P59.

These are our top twenty best selling books based on mail order and shop sales during July, August and September 1986. Our own magazines and publications are not included. The Maplin order code of each book is shown together with page numbers for our 1987 catalogue. We stock over 250 different titles, covering a wide range of electronics and computing topics.

6-Channel Burglar Alarm Panel



by Dave Goodman

This six channel alarm panel, being available ready-built, is the latest edition to the Maplin range of security systems. The panel has comprehensive facilities making it suitable for most security applications and supports sensors and detectors found in Maplin's catalogue.

Channels are switched in or out of operation from the front panel mounted toggle switches (ON = up, OFF = down), with the key switch set to OFF. Any switches that are changed after the key switch is set to ON will cause the appropriate green status LED to come on (or off!) and the red channel status LED will operate (see Figure 1). Alarm outputs will be active after delay time-out and remain active until the AUTO STOP has timed out or the key switch is set to OFF.

The hinged front panel is tamper switched and can be disconnected by the 24 hour N.O. channel 6 switch before being armed.

Figure 2 shows the internal distribution terminals and their associated circuits. Channels 1 to 5 are 'break action' circuits and detect any switch that is opened. All switches connected to an individual channel should therefore be

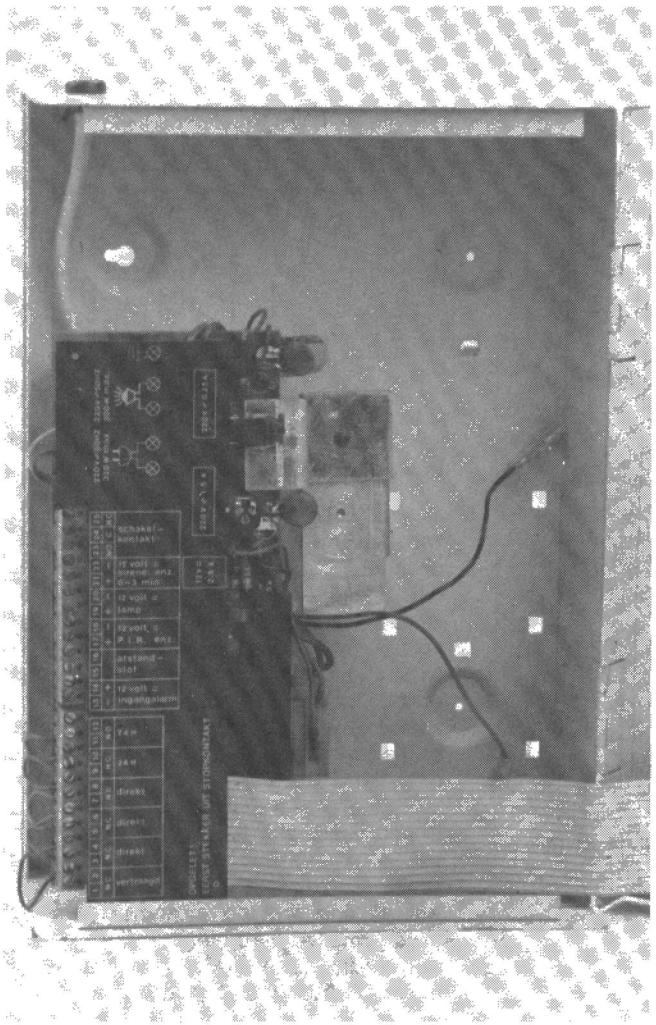
- ★ **Six Switch Selectable Channels.**
- ★ **Normally Open and Normally Closed Loops.**
- ★ **Entry/Exit and Alarm Output Auto Stop.**
- ★ **Channel Status and Memory Indicators.**
- ★ **220/240V AC Mains and Battery standby.**
- ★ **Switched Outputs supplying 12V DC and 240V AC.**

SPECIFICATIONS -
Control Panel

6 Switched Channels	5 for N.C. (Normally Closed) loops, 1 for N.O. (Normally Open) loop
Channel 1	Exit and Entry delay timed
Channel 2, 3, and 4	Exit delay timed, instantaneous entry
Channel 5	24 hr. N.C. loop, no delay
Channel 6	24 hr. O.C. loop, no delay
	Front Panel Tamper status

Zone status and memory indicators on all 6 channels. Each channel selected in or out from front panel switches. AC power on indicator and DC (battery) indicator.

Exit/Entry Indicator	On, during exit delay period Flashing during entry delay period
Exit Timer	Preset adjust, 5 sec. to 3 min.
Entry Timer	Preset adjust, 0 to 3 min.
Auto Stop	Preset adjust alarm time, 0 to 3 min.
Int. Buzzer Mute plug	Stops buzzer during Exit/Entry periods
Exit/Entry Alert	Constant +12VDC/50mA switched O/P during Exit delay period or 350ms pulsed O/P during Entry delay period
Internal Key Switch	External key switch can be fitted
PSU Output	Constant 12V DC, 0.4A supply for external sensors
Alarm Output	Switched supply 12V DC, 0.45A reset by key switch only
Alarm Output	Switched supply 12V DC, 2A reset from key switch and Auto Stop timer
Contact Alarm Output	Open, common & closed relay contacts rated at 12V DC (max 2A), reset from key switch and Auto Stop timer
Mains Input	220/240V AC @ 320W max
Mains Output	Switched 220/240V AC mains out @ 300W max, reset from key switch and Auto Stop timer
Standby battery	12V, 1.2/1.9Ah sealed lead acid, trickle charged (ave. 25mA). Not supplied with panel (see Parts List)



Inside the box.

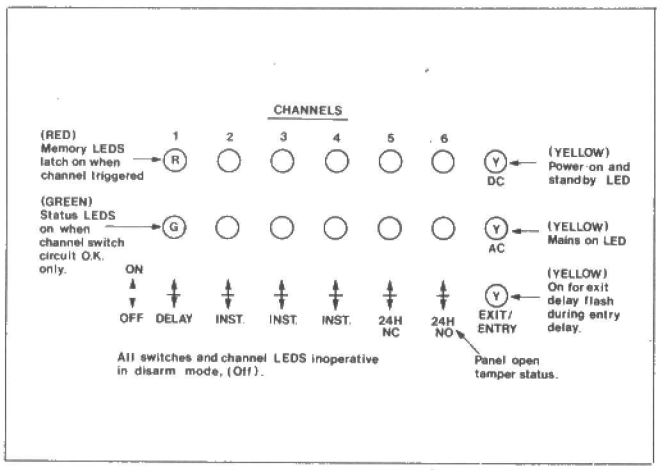


Figure 1. Control panel.

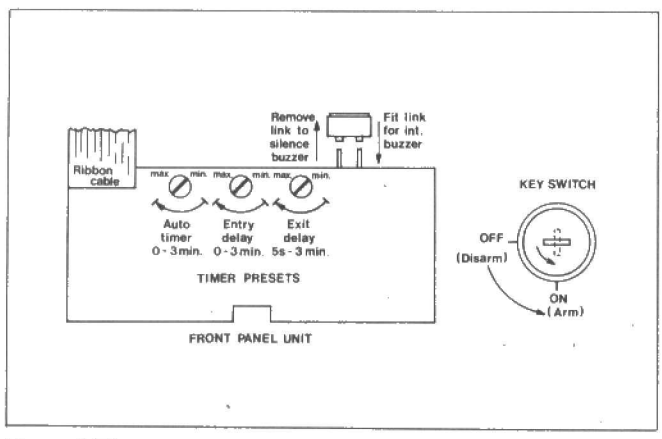


Figure 3. Timers.

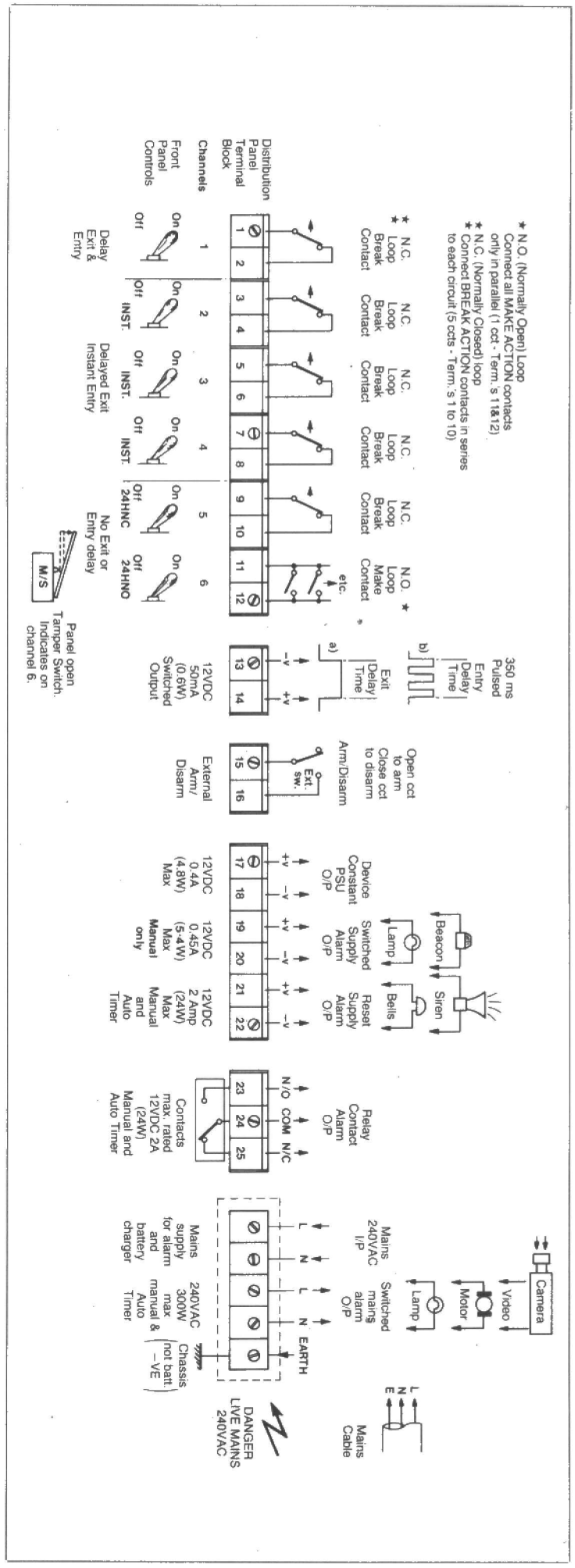

★ N.O. (Normally Open) Loop Connect all MAKE ACTION contacts only in parallel (1 cct - Term.'s 11&12)
★ N.C. (Normally Closed) loop ★ Connect BREAK ACTION contacts in series to each circuit (5 ccts - Term.'s 1 to 10)
Distribution Panel Terminal Block
Channels
Front Panel Controls
Delay Exit & Entry
Delayed Exit Instant Entry
No Exit or Entry delay
M/S
Panel open Tamper Switch. Indicates on channel 6.
★★ N.C. Loop Break Contact
★ N.O. Loop Make Contact
etc.
12VDC 50mA (0.6W) Switched Output
a)
b)
350 ms Pulsed Entry Delay Time
Exit Delay Time
External Arm/ Disarm
Ext. sw.
Open cct to arm Close cct to disarm Arm/Disarm
12VDC 0.4A (4.8W) Max
Device Constant PSU O/P
12VDC 0.45A (5.4W) Max Manual only
Switched Supply Alarm O/P
12VDC 2 Amp (24W) Max Manual and Auto Timer
Reset Supply Alarm O/P
Lamp
Beacon
Siren
Bells
Contacts max. rated 12VDC 2A (24W) Manual and Auto Timer
N/O COM N/C
Relay Contact Alarm O/P
Mains supply for alarm and battery charger
240VAC 300W max manual & Auto Timer
Chassis (not batt. -VE)
L N L N EARTH
Mains 240VAC I/P
Switched mains alarm O/P
Camera
Video
Motor
Lamp
DANGER LIVE MAINS 240VAC
Mains Cable
E N L

Figure 2. Distribution.

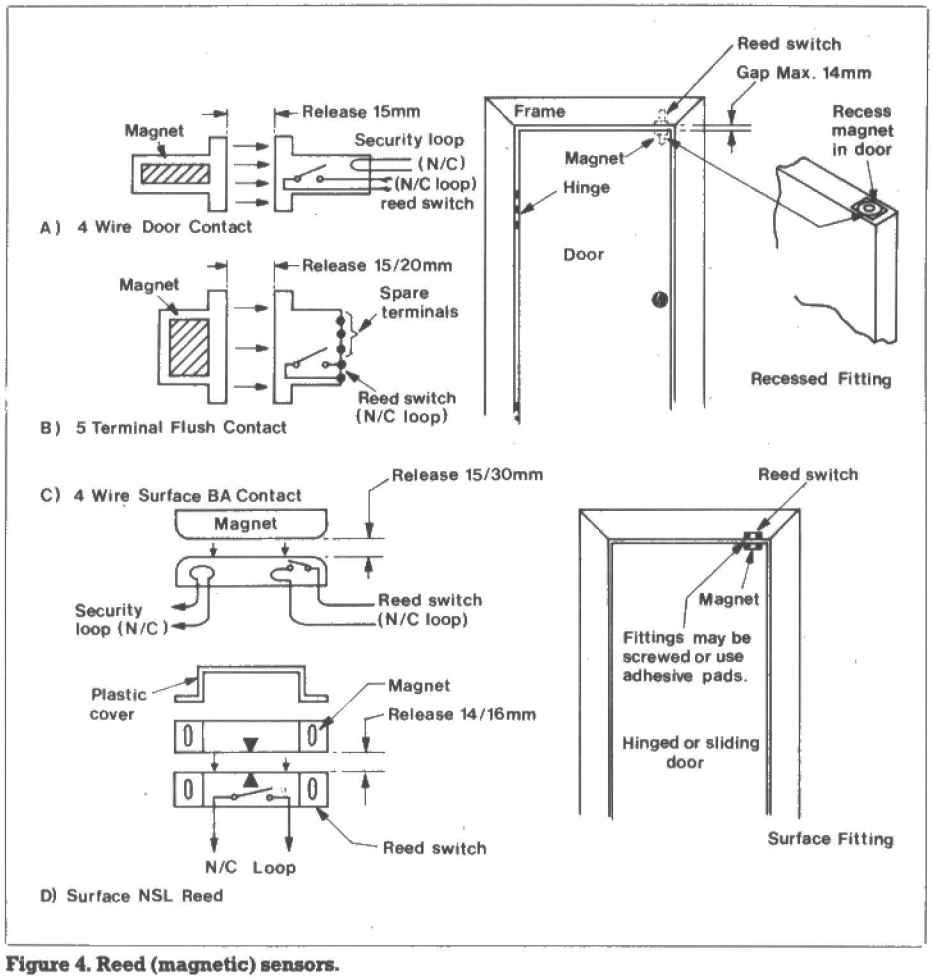


Figure 4. Reed (magnetic) sensors.

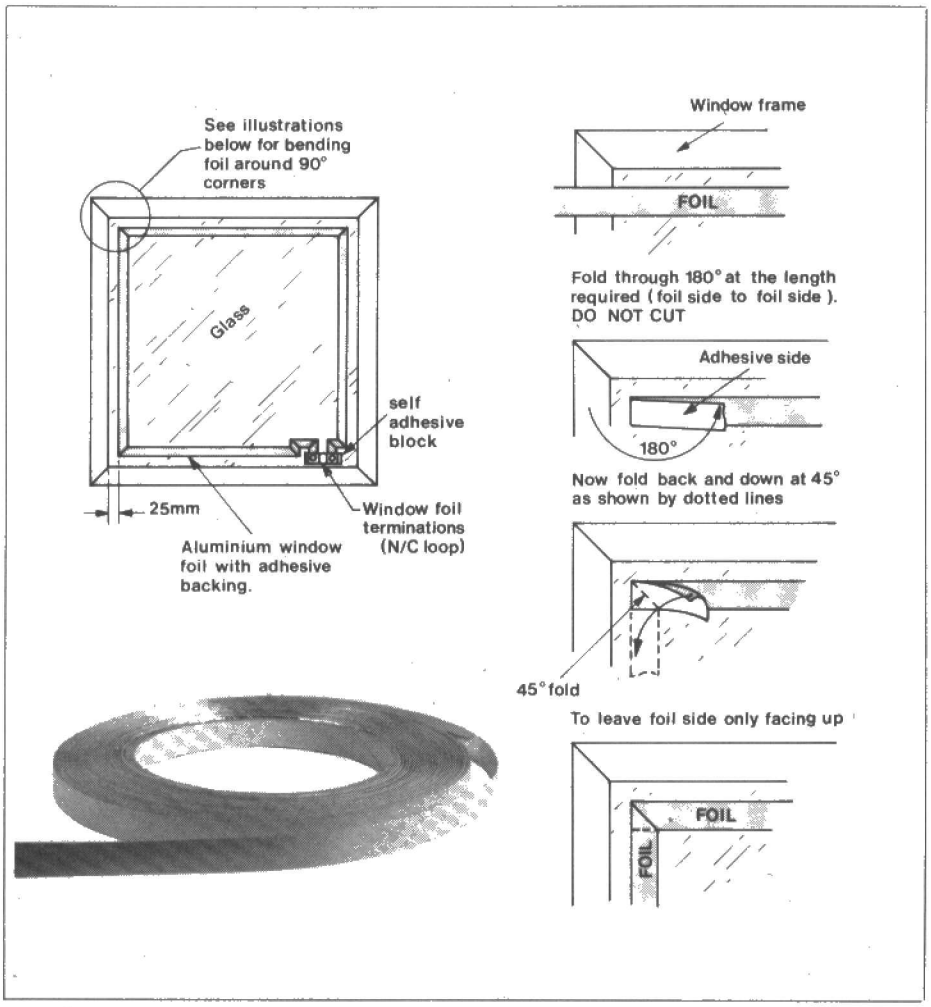

See illustrations below for bending foil around 90° corners
Glass
self adhesive block
25mm
Window foil terminations (N/C loop)
Aluminium window foil with adhesive backing.
Window frame
FOIL
Fold through 180° at the length required (foil side to foil side). DO NOT CUT
Adhesive side
180°
Now fold back and down at 45° as shown by dotted lines
45° fold
To leave foil side only facing up
FOIL
FOIL

Figure 5. Window foil.

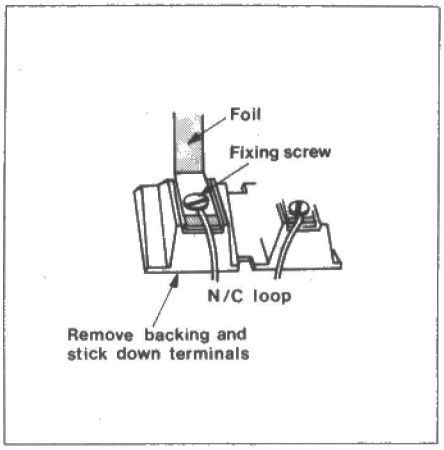


Figure 6. Foil terminations.

wired in series with each other except on channel 6. This channel detects circuits that are closed, therefore all switches should be connected in parallel with one another and not in series! During delayed exit and entry periods, the internal buzzer sounds; a separate output is available for driving external buzzers or LED's providing 12V @ 50mA during this time, see Figure 3.

External key switches can be wired in and used with the main panel switch set to ON. Various types of switch are available. Alternatively, electronic keyboards and locking systems could be used provided their output is open circuit in the ARM (ON) mode and short circuit in the DISARM (OFF) mode.

Sensors

Figure 4 shows four magnetic sensors and typical door applications. 4 wire sensors have a break action contact (door open) for use on N.C. loops and a simple 2 wire tamper loop, also for N.C. loops. The tamper loop does not have to be used. Figures 4c and 4d show surface mounted sensors which can be screwed or stuck to panels, metal and UPVC frames (particularly sliding patio doors!). On no account should UPVC or thermal break aluminium windows/doors be drilled and screwed. Use self-adhesive pads in these situations. Magnetic sensors are activated by removal of the holding magnetic field. The distance between contacts should be kept close (less than 10mm if possible) to hold contacts made and movement for breaking the contact should be 20mm or more.

Glass Sensors

Figures 5 and 6 show window foil applications. The foil is made from thin aluminium and has a self-adhesive backing for sticking on glass panels. Connections are made to both ends of the foil using a special terminal block. If the glass panel is cracked, the foil will fracture breaking the N.C. loop. Alternatively, Figure 7 shows a glass break sensor which can be fitted to N.C. or N.O. loops. The sensor should be mounted at least 75mm (3 in.) away from frames and stuck onto glass panels with the self-adhesive disk supplied.

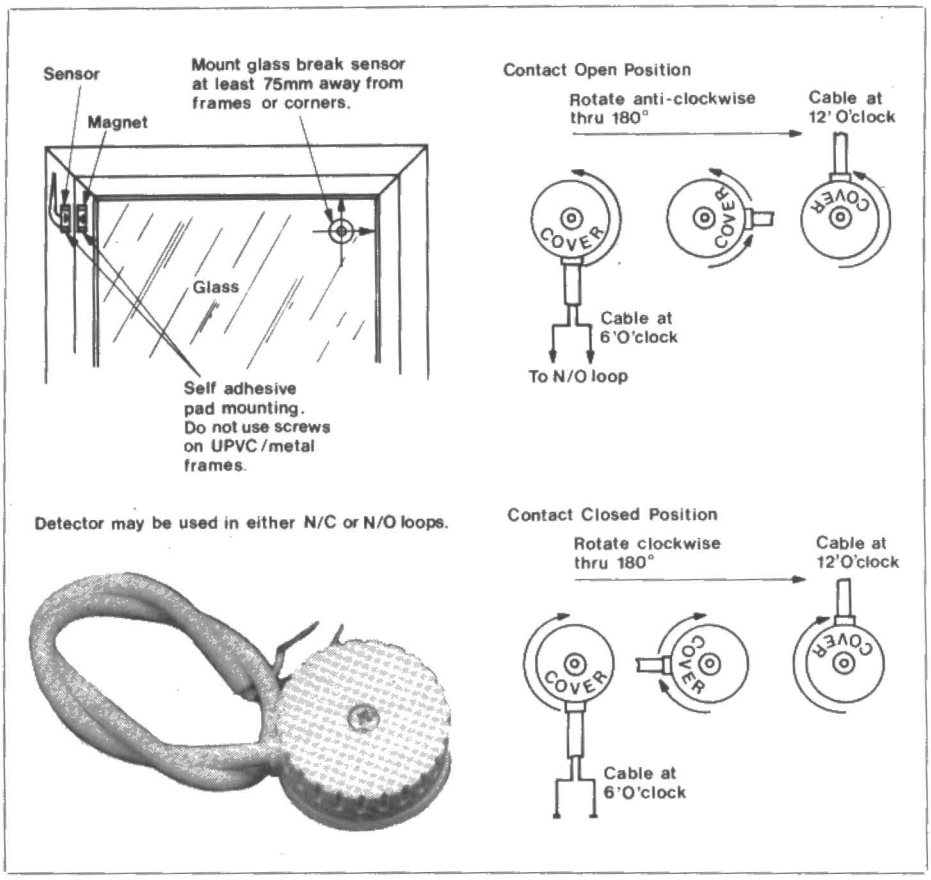

Sensor
Mount glass break sensor at least 75mm away from frames or corners.
Magnet
Glass
Self adhesive pad mounting. Do not use screws on UPVC/metal frames.
Contact Open Position
Rotate anti-clockwise thru 180°
Cable at 12' O'clock
COVER
Cable at 6'O'clock
To N/O loop
Detector may be used in either N/C or N/O loops.
Contact Closed Position
Rotate clockwise thru 180°
Cable at 12'O'clock
Cable at 6'O'clock

Figure 7. Glass break detector.

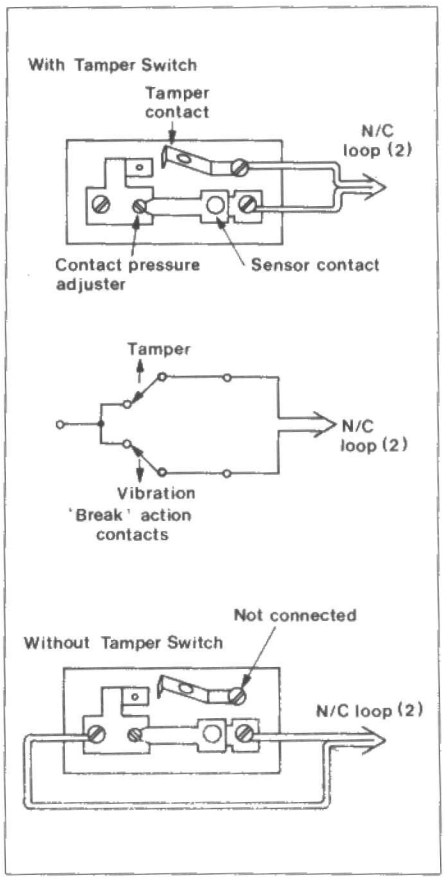


Figure 8. Vibration sensor.

Vibration Sensor

This device has an adjustable pressure contact which responds to sharp, sudden movements. The unit could be fitted to glass panels, but wind or vehicles could falsely trigger the sensor. Partition panels or walls can be protected from violent entry or stair risers could be sensed with this device. Figure 8 shows two connection methods, one using the tamper switch, activated by removing the cover and the other without tamper protection for N.C. loops.

Heat Sensor

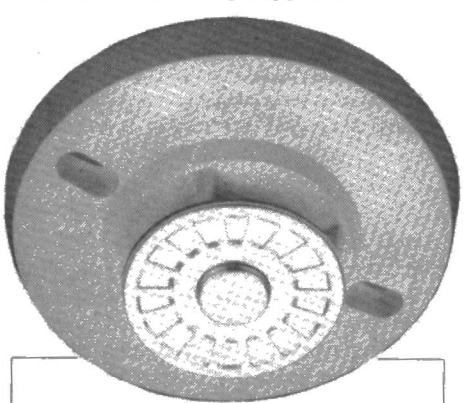
The heat sensor shown in Figure 9 is a N.O. loop system which becomes active at temperatures at and above 60°C. The release temperature is at approximately 40°C and below. The sensor should be mounted high up on ceilings, as hot air rises, and is most useful in areas likely to be a fire hazard e.g. kitchen and airing cupboard.

Panic

Both panic and help switches, see Figure 10, should be connected to N.O. loops where immediate activation is necessary. Ideal areas for mounting these switches are close to main entrance areas or by the bed.

Pressure

Two 4 wire pressure mats are available for N.O. loops. The extra security loop is for use on N.C. loops, see Figure 11, although it is not necessary to use this. A small, narrow mat fits under carpets on stair treads, whereas the larger version is placed beneath a rug or carpet in front of doorways or windows.



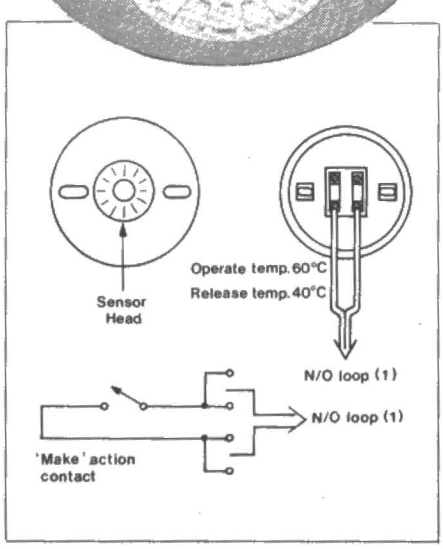


Figure 9. Heat sensor.

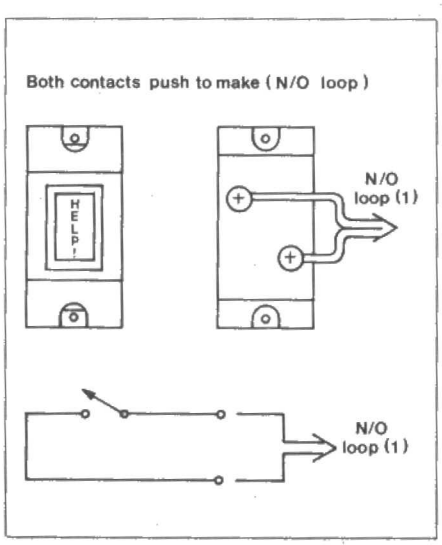


Figure 10. Help button.

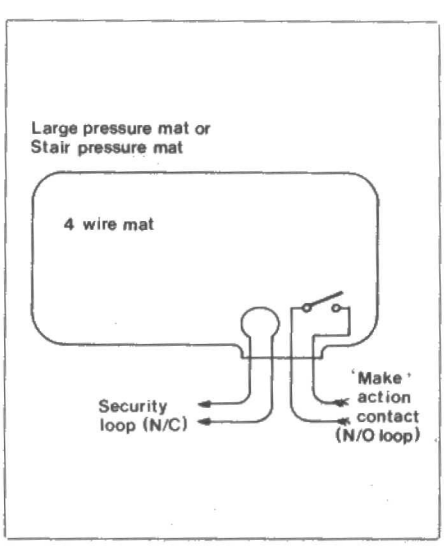


Figure 11. Pressure mat.

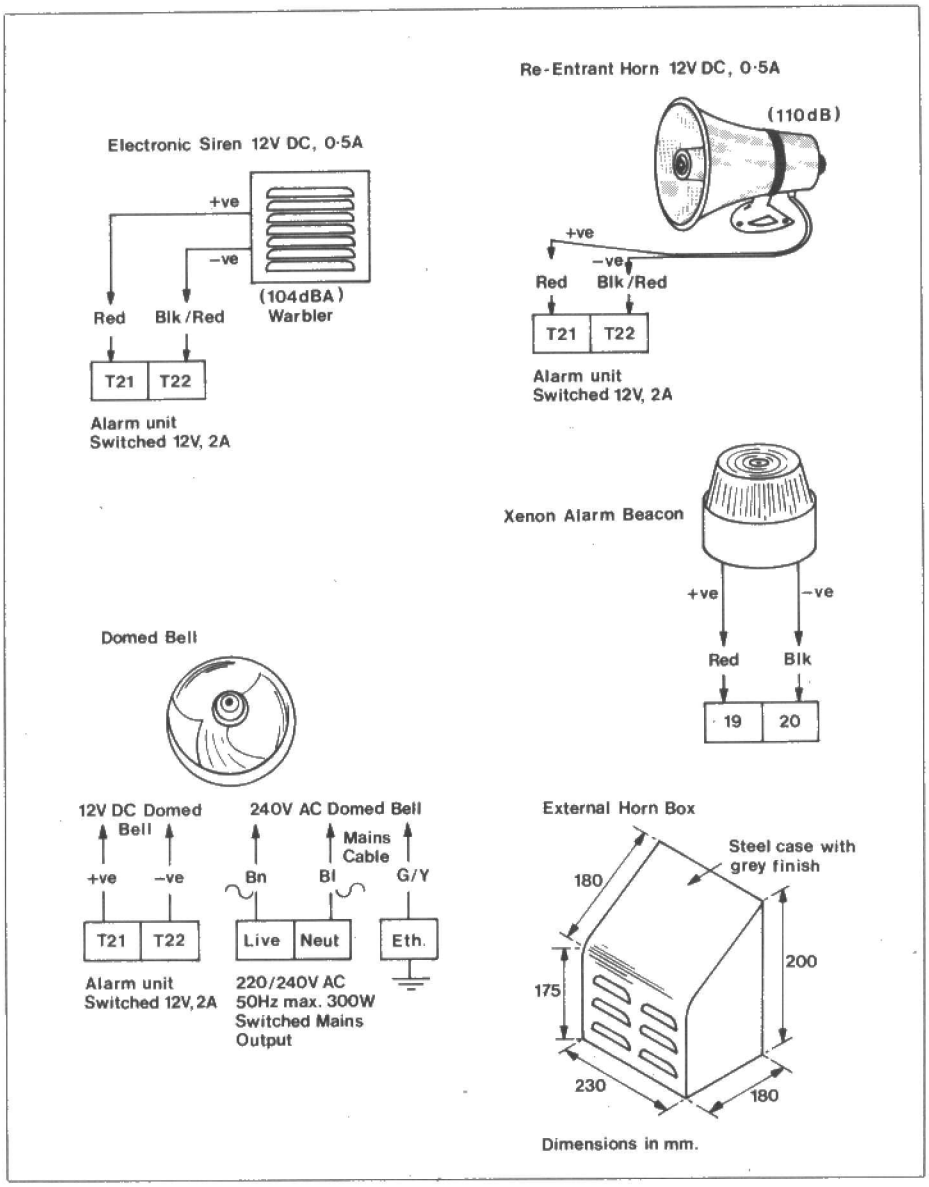


Figure 12. External indicators.

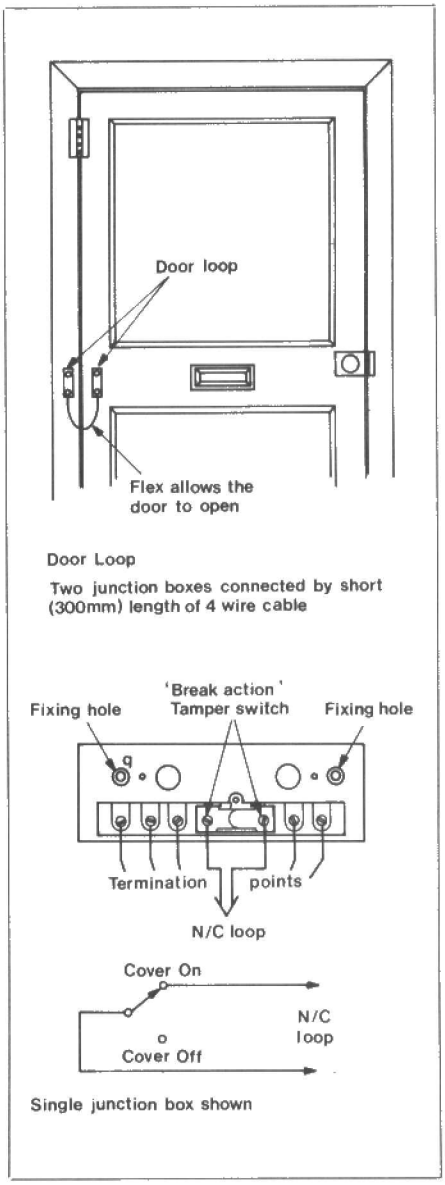


Figure 13. Door loop.

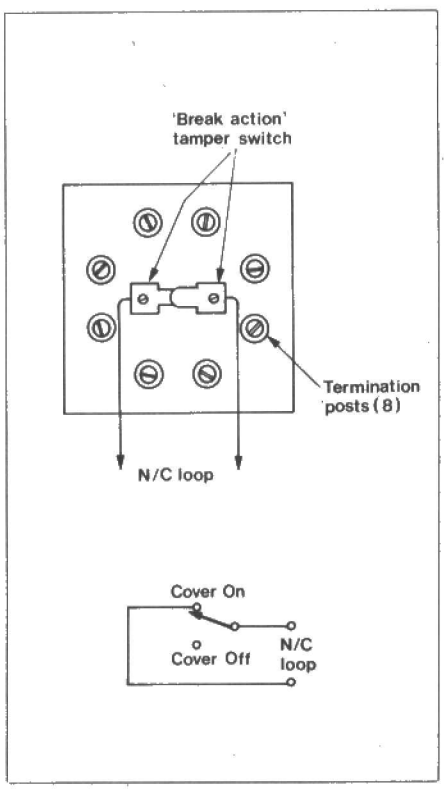


Figure 14. Junction box.

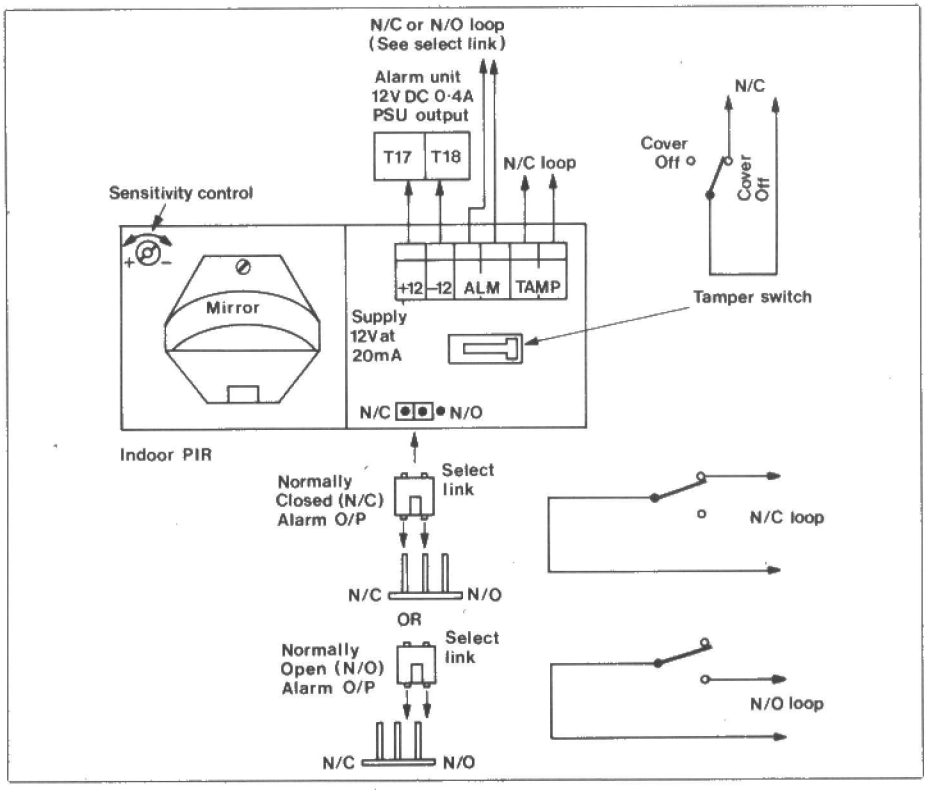


Figure 15. Infra-red detector.

Alarm Indicators

Several types of indicator are shown in Figure 12. The traditional type of domed bell, available in both 12V DC and 240V AC versions can be driven from the alarm panel and should be connected to Auto timed outputs. The same applies to the electronic siren and re-entrant type horn - both of which are extremely loud. In domestic situations, alarm sounders should be turned off after 17 minutes and indication continued with a flashing beacon. Fit the beacon to a manual reset output. If using dry relay contacts on the panel to switch external horns and batteries, then the external horn box can be used to accommodate these items.

The box is made of steel, finished in grey. Louvres punched into the front allow sound out and a beacon could be fitted onto the top panel. Dimensions as shown in the drawing.

Junction Boxes

A single 5-terminal junction box and a flexible door-loop using two 5-terminal boxes is shown in Figure 13. These units offer a convenient method of connecting sensors to a common loop. The larger unit shown in Figure 14 has eight termination points. All three versions have an integral N.C. tamper contact.

Movement Detectors

The PIR detector in Figure 15 senses body heat in the infra-red band and will trigger either N.C. or N.O. loops. These systems are not normally affected by spurious inputs, but precautions should be taken to prevent direct sunlight from heating the detector. Alternatively, a doppler-radar detector, see Figure 16, is available in kit form. This unit has a wide detection area and any object moving in its field will trigger the N.O. loop output (details in Project Book Three).

Standby Supply

The alarm panel has mains power fail battery back up facilities, requiring the addition of a sealed, lead acid 12V battery. Two types of battery are available in either 1.2Ah or 1.9Ah versions. The larger capacity battery will run the system for much longer periods than the smaller version and this fact should be kept in mind when deciding on external accessories and their power requirements, see Figure 17.

Be Secure

Armed with this knowledge and with the aid of the detailed diagrams shown, you should be able to secure your property from unwanted intruders. Figure 18 gives typical positioning for various sensors. The 6-Channel alarm panel on which this article is based is a new item in the 1987 Maplin catalogue, so rush out and get your copy now. Some of the sensing devices are new too. Why pay hundreds to have a system installed when you can do-it-yourself?

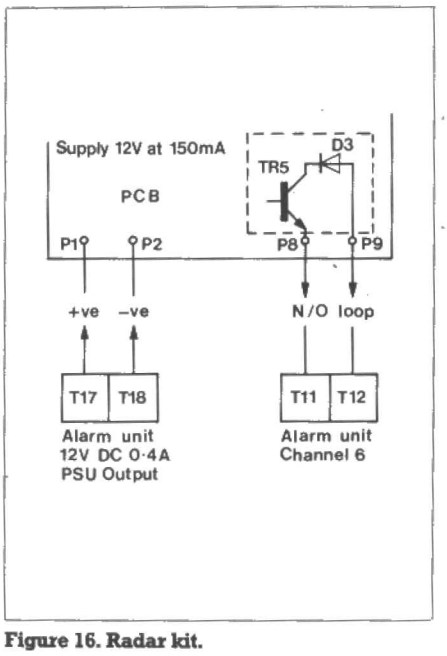


Figure 16. Radar kit.

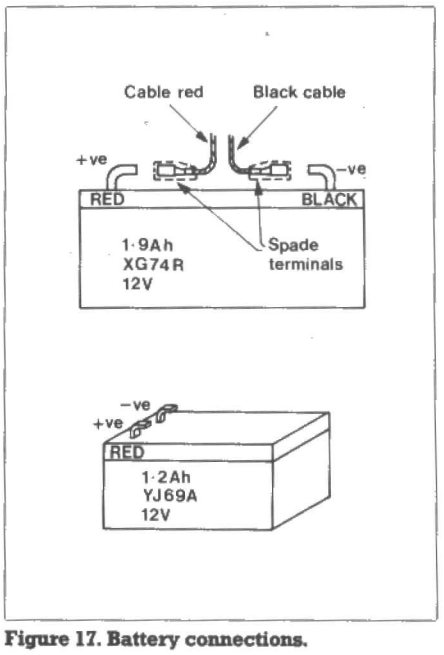


Figure 17. Battery connections.

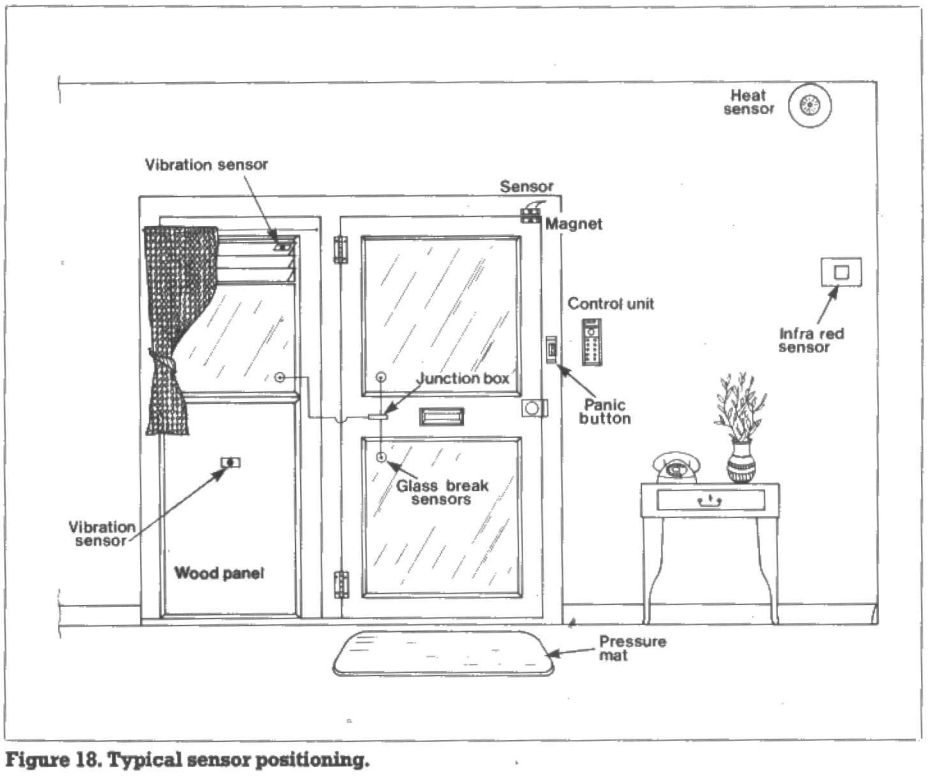
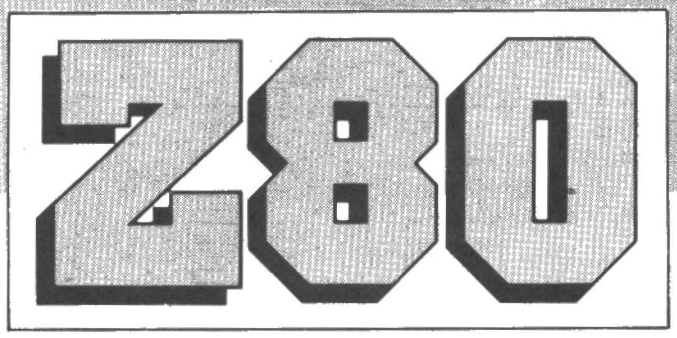


Figure 18. Typical sensor positioning.

PARTS & ACCESSORIES

Description	Stock Codes
6-Channel Alarm Panel	YN57M
Key Switch	FH40T
Plastic Key Switch	FV42V
Round Key Switch	FV41U
4-Wire Door Contact	YW46A
5-Terminal Flush Contact	FK77J
4-Wire Surface BA Contact	YW47B
Surface NSL Reed	FP10L
Window Foil	YW50E
Window Foil Terminals	YW51F
Glass Break Detector	FP11M
Vibration Sensor	FK78K
Heat Sensor	FK47B
Panic Button	FK46A
Help Button	FP12N
Door Loop	YW48C
Single Junction Box (5-Way)	YW49D
8-Way Junction Box	FK76H
Large Pressure Mat	YB91Y
Stair Pressure Mat	FK79L
Electronic Siren	XG14Q
Metal Horn 5-Inch	YN59P
12V DC Domed Bell	YK85G
240V AC Domed Bell	YK58N
Xenon Alarm Beacon	YK39N
External Box	XG07H
Signal/Burglar Alarm Cable (4-Wire)	XR89W
100m 4-Wire Burglar Cable	PA77J
Indoor PIR	YM87U
RTX3 Radar Doppler Unit (Kit)	LW73Q
Project Book 3	XA03D
1.2Ah L/Acid Batt. 12V	YJ69A
1.9Ah L/Acid Batt. 12V	XG74R

MACHINE CODE PROGRAMMING WITH THE Z80



by Graham Dixey C.Eng., M.I.E.R.E. Part Five

The first half of this series on the Z80 has covered the instruction set, the addressing modes available, plus a few application examples. There is little doubt that the best way to become a proficient machine code programmer is by constant practice. This will develop a greater degree of familiarity with the areas mentioned above and also increase greatly one's self-confidence. The remaining parts of this series will attempt to show how to develop programs to implement certain ideas. Included later will be a discussion of the use of the Z80 and its PIO and CTC chips, and also the thorny problem of 'interrupts'. But more of that at the appropriate time. For now the general approach to writing short programs, which may merely be segments of larger programs, will be dealt with. It is a very good idea to develop a systematic method of dealing with program design.

Modular Programming

In some respects program design is like circuit design. You can just start at one end and let your creation grow and grow until it does what it's supposed to. Perhaps later parts of the design will link back to earlier parts and perhaps some elements will appear more than once. In the case of circuit designs on breadboards, the result is often the classical 'bird's nest', which may not even work when it is later rebuilt on a proper circuit board!

Modular design, as far as programming is concerned, means looking at the overall view of what the program is intended to achieve and then breaking it down into a number of discrete functions. Each of these is termed a 'module' and can be designed quite separately; it is then 'linked' to other modules with which it is intended to interact. Each module is capable of being tested quite separately and this has the advantage that, when a bug appears (there's bound to be at least one in a program of any length!), it's usually possible to pin-point the module in which the bug lies and work on that module to de-bug it. Initial considerations of what the modules are intended to do will also identify repeated use of a particular module, leading to the use of sub-routines. The latter shorten the program and save memory space.

Take an example of a keyboard input routine. The problem is stated first and then broken down into modules, including sub-routines.

Pressing a key generates a code for that key, this code being unique to that type of keyboard (i.e. not ASCII, etc); as well as identifying which key has been pressed in a general sense, it is required to know whether the CONTROL key or the SHIFT key was pressed at the same time; the code, when identified, has to do two things - first it must display the digit, corresponding to the keypress, on a seven-segment display, and secondly it must store that digit in memory as its HEX value.

How many modules can you identify from this description, and how many of them will be sub-routines?

A first consideration may indicate that there are FIVE modules, as follows:

Module 1: Inputs data from the keyboard and tests for CONTROL or SHIFT keypress.
Module 2: Converts the keyboard code to a code suitable for driving seven-segment displays.
Module 3: Sends the code to the display.
Module 4: Converts the keyboard code to HEX code.
Module 5: Sends the code to memory.

However, further consideration may show that some of the above modules are too trivial to be classed as separate modules, an example being Module 5. The above initial breakdown is given to show how one's first thoughts may run and how it is possible to overdo even good ideas.

So Module 4 becomes 'converts the keyboard code to HEX and sends it to memory'.

It may be wondered why Modules 2 and 3 have not been combined in the same way. Sometimes they could be but the chances are that the routine to update a strobed multi-digit 7-segment display will be more complex than merely storing a HEX code in memory. Not only this, but the routine to send data to the display may become a sub-routine to be called any time that the display is to be updated. Thus, of the four modules now left, Module 3 is a sub-routine. The sequence of events may now appear as follows.

1. Input a character from the keyboard and test it.
2. Convert it to seven-segment code.
3. CALL the display sub-routine.
4. Convert the keyboard code to HEX and store it.

Look at these four lines again and note that lines 2 and 4 execute operations that it is easy to visualise will be needed over and over again. Could they not also be sub-routines? Obviously if they were, there would then be one input module (Module 1) and three sub-routines. There's nothing wrong with this; often programs consist of little else but a collection of sub-routines called again and again by the main program. However, if the conversion to seven-segment code is always associated with a subsequent conversion to HEX, then both conversions can be combined into a single sub-routine, which might be called the 'convert' sub-routine or whatever. To preserve its order within the program (assuming this is necessary), it can have the 'display' sub-routine 'nested' within it. The sequence now becomes:

1. Input a character from the keyboard and test it.
2. CALL the 'convert' sub-routine, which develops the required seven-segment code.

3. CALL the 'display' sub-routine, which updates the display; RETURN to the 'convert' sub-routine.
4. Develop the appropriate HEX code and store it. RETURN to the main program.

Notice from this how the 'display' sub-routine is nested within the 'convert' sub-routine. This is shown diagrammatically in Figure 1.

Now consider how to write some of the modules in this program because, although it's useful to discuss general ideas, it is of greater value still to the relative beginner to machine code to see how programs are developed from initial ideas.

First consider the input module. This must accept and examine the codes from the keyboard. Before anything can be done in the way of designing this module, it is necessary to know in detail the nature of the keyboard codes. Assume that, for this particular keyboard, the codes follow the form that 0 is represented by &C0, 1 is represented by &C1, 2 is represented by &C2, and so on. The reason for these particular codes is not important; they do actually exist for a known type of keyboard. All that matters is that when the numeric key 1 is pressed, the generated code &C1 is converted to the right seven-segment code and then to the HEX value for 1. At the same time a check is made to see if this keypress was accompanied by a CONTROL keypress or a SHIFT keypress. The latter are identified as follows. If the CONTROL key is pressed, 'bit 7' of the keyboard code becomes '0', but if the SHIFT key is pressed, 'bit 6' becomes '0'. This may sound complicated but isn't really, as an example will show.

Thus, the keyboard code for '1' is &C1, or 11000001 in binary. And, since bit 7 is the most significant bit, the code for 'CONTROL plus 1' is 01000001 in binary, or &41; similarly when 'SHIFT plus 1' occurs bit 6 goes low, giving 10000001 in binary, which is equivalent to &81.

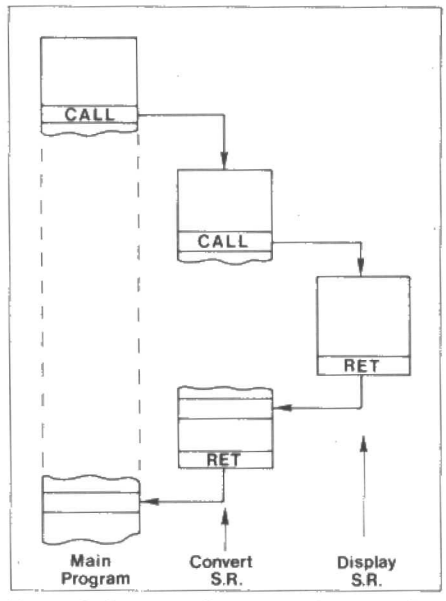


Figure 1. An example of nested sub-routines.

Summing up then,

Numeral 1 = &C1;
CONTROL + 1 = &41;
SHIFT + 1 = &81.

The question then is, 'how is it possible in a simple way to identify each of these three cases for ALL possible input characters? The answer must depend on whether bits 7 and 6 can be tested to find whether they are high or low. As it happens, the BIT instruction allows us to test any bit within a byte in a specified register. Look at the following program segment.

```
IN    A,(DATA)  Get character from keyboard
BIT   7,A       Test bit 7 and,
JP    Z,CTRL    if zero go to CTRL
BIT   6,A       Test bit 6 and,
JP    Z,SHIFT   if zero go to SHIFT
CALL  CONVERT   Otherwise convert keyboard code
```

The keyboard is an external device and the IN command is used to fetch data from it into the A register; it has the label DATA and is treated as a 'port' with an address which depends upon the design of the system but is assumed to be &00 in this particular case. The first BIT instruction tests bit 7 of the A register and sets the zero flag if bit 7 is found to be zero. This allows the next line to jump 'conditionally' to a label called CTRL (some routine that handles CONTROL inputs) if bit 7 is zero. The same idea is used in the next two lines to test bit 6 and go to a label SHIFT. However, if neither bit 7 nor bit 6 was found to be zero, then neither the CONTROL key nor the SHIFT key was pressed and the program can go straight to the process of converting the keyboard code, which it does by calling the 'convert' sub-routine mentioned earlier. The assembled version is found in Listing 1, which includes the 'assembler directives' EQU (telling the assembler the port address for the keyboard) and ORG (telling the assembler where to store the code).

It's worth also having a look at how the convert sub-routine works. The problem is this; if a keyboard code is, for example, &C1 representing numeral 1 as agreed, how can this easily be converted to the code to drive a 7-segment display which is &18 (assuming common-cathode displays), and then converted to &01, which is the HEX equivalent?

The answer is to use a 'look-up table', in which the codes are stored in sequence so that they can be accessed by using the appropriate addressing mode. The following shows the first four lines of a look-up table for 7-segment codes.

TABLE ADDRESS	CHARACTER	7-SEGMENT CODE
&01C0	0	7B
&01C1	1	18
&01C2	2	B3
&01C3	3	BA

and so on.

That's enough of the table to show the general principle. It's been made particularly easy in this case by the choice of what is called the 'base address', which is &0100. Note that the

```
ARNOR Z80 ASSEMBLER  version 1.13

00002  5C00  (5C00)              ORG   &5C00        ;Define start address
00003  5C00  (0000)    DATA      EQU   &00         ;Define keyboard address

00004  5C00  (5D00)    CTRL      EQU   &5D00       ;Define address,CTRL
00005  5C00  (5E00)    SHIFT     EQU   &5E00       ;Define address,SHIFT
00006  5C00  (5F00)    CONVERT   EQU   &5F00       ;Define address,CONVERT
00007  5C00  DB 00               IN    A,(DATA)    ;Fetch keyboard data
00008  5C02  CB 7F               BIT   7,A         ;Test bit 7 and,
00009  5C04  CA 00 5D            JP    Z,CTRL      ;if zero go to CTRL
00010  5C07  CB 77               BIT   6,A         ;Test bit 6 and,
00011  5C09  CA 00 5E            JP    Z,SHIFT     ;if zero go to SHIFT
00012  5C0C  CD 00 5F            CALL  CONVERT     ;Or convert keyboard code
```

Listing 1. Keyboard input routine testing for CONTROL and SHIFT keypresses.

```
ARNOR Z80 ASSEMBLER  version 1.13

00002  3A00  (3A00)              ORG   &3A00          ;Define start address
00003  3A00  (5B00)       DISP   EQU   &5B00          ;Define address DISP
00004  3A00  11 00 01            LD    DE,&0100       ;Base address,7seg. table
00005  3A03  F5                  PUSH  AF             ;Preserve keyboard code
00006  3A04  5F                  LD    E,A            ;DE holds table pointer
00007  3A05  1A                  LD    A,(DE)         ;Get 7seg. code
00008  3A06  02                  LD    (BC),A         ;Send code to display RAM
00009  3A07  CD 00 5B            CALL  DISP           ;Call display sub.
00010  3A0A  11 20 01            LD    DE,&0120       ;Base address, HEX table
00011  3A0D  F1                  POP   AF             ;Return keyboard code
00012  3A0E  5F                  LD    E,A            ;DE holds table pointer
00013  3A0F  1A                  LD    A,(DE)         ;Get HEX code
00014  3A10  C9                  RET
```

Listing 2. The DISPLAY sub-routine.

table is accessed merely by adding the keyboard code to the base address to get the right table address.

Thus, when the keyboard code is &C2, this is added to &0100 to get the table address &01C2, at which the 7-segment code for 2, namely B3, is found. A similar process is used to get the HEX code afterwards, merely by using a different base address for the HEX codes table. Listing 2 shows the assembled version of the conversion sub-routine, which works as follows.

The first line loads the DE register pair with the base address of the 7-segment look-up table, which in this example is &0100. The contents of the A register are pushed onto the stack in the second line because they will be needed again for the HEX conversion; PUSH AF is used because, although there's no need to push the flags register as well, there isn't any choice because the Z80 doesn't have an instruction that just pushes A - it's of no consequence. The keyboard code held in A is added to the base address not by using an addition operation as you might think but simply loading the E register from A; D holds &01 as before but E holds the keyboard code, the two together forming a 'pointer' to the appropriate memory location in the look-up table. Line 4 fetches the required 7-segment code into the A register by loading A from (DE); the latter, you may remember, meaning 'the address pointed to by DE'. Next an address pointed to by the contents of the BC register pair has this 7-segment code stored in it. This address is one of a short block of memory whose function is to hold the data for the strobed multi-digit display. It's not anything to worry about at the moment. To update the display the sub-routine, known simply as DISP for brevity, is called. After returning from this sub-routine, the next task is to convert the keyboard code to HEX for storage in some memory location. To access the look-up table for HEX codes, the DE register pair is loaded with a new base address, in this case &0120. However, the keyboard code in the A register has been overwritten (which is why we pushed it earlier) so we must now 'pop' the stack to get it back. As with the other conversion the keyboard code is added to the base address by loading the code into the E register. The next load operation, LD A, (DE), fetches the HEX code from the address in the look-up table pointed to be DE. Finally, a RETurn is made from the sub-routine.

Code Conversions

The previous example included methods of converting from a fairly obscure keyboard code into 7-segment and HEX codes. It is often useful to be able to convert between common codes, such as ASCII, BCD and HEX, and look-up tables are not always necessary for such conversions. As a very simple example, consider the conversion of ASCII to BCD and vice-versa. BCD, by definition, only uses the ten digits 0 to 9 so it is only necessary to convert part of the ASCII set. Examination of an ASCII table shows that the digits 0 - 9 are represented by the codes &30 to &39 respectively; in binary &30 is represented by 00110000 and &39 is represented by 00111001 and obviously the digits in between follow the same pattern in which the left hand nibble is always 0011 (3) and the right hand nibble is the four-bit BCD group for the corresponding digit, e.g. 0000 for 0, 0001 for 1,.....1001 for 9. All that is needed is to get rid of the left hand nibble.

This can be done by using a standard technique involving the use of the AND instruction. The basis of the AND operation is that, in AND logic, anything ANDed with 1 equals itself, while anything ANDed with 0 becomes 0. Thus, to reduce the left hand nibble to zero while preserving the right hand nibble, all that has to be done is to AND the former with 'all zeros' and the latter with 'all ones', i.e. the word is ANDed with &0F, as follows:

```
LD    A,ASCII   Load A register with ASCII code
AND   &0F       AND A register contents with &0F
```

The A register will now contain the BCD equivalent of the ASCII code.

Is there any problem in reversing the operation, in changing a BCD number into its ASCII equivalent? Suppose the BCD number was 8, i.e. 00001000, what would have to be done to it in order to change it into ASCII? Since the ASCII code for 8 is 38, i.e. 00111000, the answer is, that the left hand nibble would have to assume the value 3, i.e. 0011. The instruction that allows us to 'add in' bits to a word is the logical OR. The basis of the OR operation is simply that anything ORed with 1 becomes 1. Thus ORing 0000 with 0011 produces 0011, which is just what is wanted. ORing with 0 changes nothing.

```
LD    A,BCD     Load the A register with the BCD code
OR    &30       OR the contents of the A register with &30
```

These two operations, that allow us to 'mask out' unwanted bits or to 'add in' bits into specific positions within a work, are very useful and much use can be made of them. Look at them again in an entirely different application.

Suppose a computer is connected through a pair of 8-bit input/output ports to some peripheral that it is controlling. It doesn't matter in the slightest what it actually is - the principle is universal. In order to do its job of control it will have to receive signals from the controlled device, that tell it what is happening and then it will have to send controlling signals to the peripheral to make it perform the required actions. Sometimes these control or data signals will be 8-bit parallel words, but often they will be individual lines that are simple switched levels, in or out. Thus, any individual bit within the word will have to become logic 1 or logic 0, according to the

situation, quite independently of any other.

There are two separate situations, one for signals input to the computer, and another for signals output from the computer. Take the former first.

If a signal is input on a single input line in an 8-bit parallel port, the problem is one of isolating it to see if it is a 1 or a 0. To do this the AND instruction is used to mask out the unwanted bits. All that is necessary to decide on the value of the 'mask word' is to know the position of the bit to be tested. Suppose that it is bit 3, then all other bits must become zero when the input word is loaded into the A register. The masking operation can be followed with a test for a zero or non-zero result, as follows:

```
IN    A,(DATA)   Fetch word from input port
AND   &08        Mask out all bits except bit 3
JP    NZ,FRED    If bit 3 is one, go to routine called FRED
```

The use of FRED as a label is not an attempt at humour; it is a means of generalising - that is, it assigns a name to a routine about which we actually know nothing (nor need to).

Convince yourself that the operand for the AND operation is correct. The HEX value &08 is 00001000 in binary; thus, it can be seen that there is a 1 in the bit 3 position only. The JP instruction is conditional upon the result of the AND operation being zero. If it is, the jump is taken to FRED; if it isn't, the program merely goes onto the next line.

Now consider a case of an output line. Suppose that several relays are connected to some of the output lines, while there are indicators, e.g. LED's connected to others. Some of the relays may be energised, some not; some indicators will be on and some off. It has been decided that one particular relay must be switched on without affecting the state of the other relays and the indicators. The answer should be obvious; a logic 1 must be sent to that bit position by using the OR instruction. Suppose it is bit 0. Then the operand for the OR operation will be &01, and the program segment is as follows:

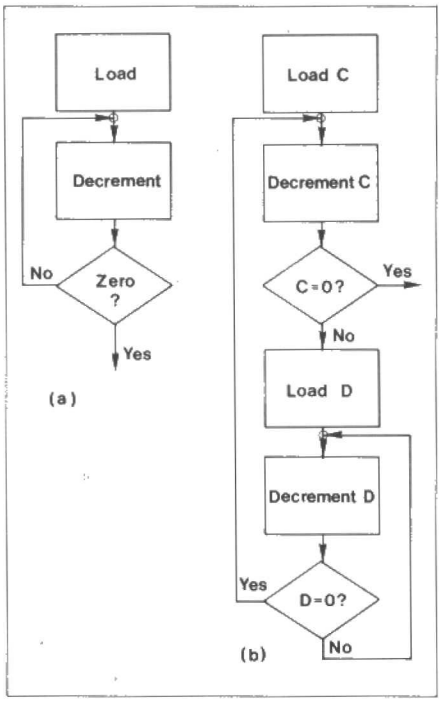


Figure 2. Flowcharts for (a) single-loop delay and (b) double-loop delay.

```
IN    A,(PORT)   Fetch current state of output port
OR    &01        OR this with &01 to force bit 0 to 1
OUT   (PORT),A   Send modified control word to output port
```

The comments should make it clear what is happening.

Obviously before anything can be done to change the state of the output port logic levels, it is necessary to fetch the word describing its current state. The logical OR operation is always carried out between the A register and some specified operand; it is not possible to carry it out at the port directly.

Finally, a program that is used to generate a given delay. It is a useful utility because it can be used to set up time intervals in programs that generate waveforms and also used to slow down events to make them visible, e.g. when operating LED's at input/output ports in control demonstrations.

The Loop and Double-Loop Delay Programs

A single loop will generate quite a short delay, up to about 260 microseconds with a 1MHz clock, proportionately less with a higher frequency clock. The double-loop program will multiply the possible delay by a factor of about 250. If you really want to get very long delays, the principle can be extended to use three loops. Figure 2 shows the flowcharts for both single and double loops. The basic principle is, that a register is loaded with an initial value (according to the delay required); this is then decremented and tested repeatedly until the register contents are found to be zero. The program then exits the loop. This describes the single delay loop, the assembler program for which is given below.

```
      LD     C,(DATA   Loads initial value
LOOP  DEC    C         Decrement register
      JR     NZ,LOOP   If not yet zero loop again
```

To extend the principle, after the first register, say register C, is decremented, the program loads another register, e.g. the D register, with some data and this is decremented and tested as for the single loop program, i.e. until the D register contents are zero. The program then jumps back to the first loop, decrements the C register, tests it for zero and, if it finds that it isn't, goes into the second loop again. In other words the single loop program is made to occur repeatedly according to the value loaded into the first register. The assembled version of this program is found in Listing 3.

It would be easy to question the choice of programs used to illustrate this article, whose purpose it was to show how to approach the writing of machine-code with some program examples. The truth, of course, is, that the field is absolutely limitless and everyone will have their own ideas about what they want to write programs for. Perhaps they will have picked up one or two ideas from the foregoing. In the next article, as was mentioned earlier, there will be an in depth discussion of the use of 'interrupts.'

```
   ARNOR Z80 ASSEMBLER  version 1.13

00002   3F00   (3F00)                ORG    &3F00        ;Define start address
00003   3F00   0E FF                 LD     C,&0FF       ;Load C with data
00004   3F02   0D            LOOP1   DEC    C            ;Decrement C register
00005   3F03   28 08                 JR     Z,FINISH     ;Leave program if zero
00006   3F05   16 FF                 LD     D,&0FF       ;Load D with data
00007   3F07   15            AGAIN   DEC    D            ;Decrement D register
00008   3F08   28 F8                 JR     Z,LOOP1      ;If zero go back to loop 1
00009   3F0A   C3 07 3F               JP     AGAIN        ;Otherwise go again
00010   3F0D   76            FINISH  HALT                ;End of program
```

Listing 3. The double-loop delay program.

TEST GEAR AND MEASUREMENTS

by Danny Stewart
Part 3

Part 2 discussed bridges, but before we abandon our bridges altogether, let us talk a bit more about them.

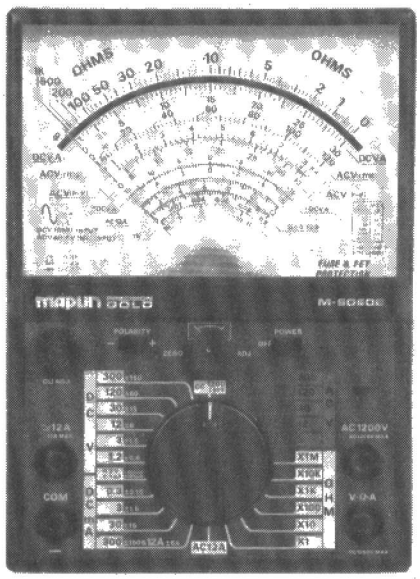
Potentiometers

Potentiometers are in fact a kind of bridge since we carry out a comparison with a known standard and look out for balance or null indication. As with bridges, a high accuracy is achieved since the potentiometer is independent of the characteristics of the null detector. The main difference between potentiometers and bridges is that potentiometers are used to measure voltages and indirectly, current. Since the detector is a null detector it does not consume power or mask the reading and therefore potentiometers are used widely for calibrating ammeters and voltmeters.

Figure 1 shows a basic slide wire potentiometer. The wire is carefully manufactured to give it uniform resistance along its length. Along the length is a metre ruler and if the wire is known to drop 1 volt per metre, then any other voltage can be measured by the sliding contact. In order to set the voltage at 1 volt per metre, the calibrate switch is thrown towards the standard cell and the rheostat adjusted to give a null reading on the galvanometer at the 1 metre mark. Assuming the standard cell is 1 volt and the resistance of the wire 100 ohms per metre, this gives a current of 10mA. Now any other cell or unknown voltage up to about 1.6V can be measured against the slide wire. It is not practical to have an instrument more than about a metre long and the resistors may be in short loops, see Figure 2. In addition, a voltage divider R_1 and R_2 and a switch may be used to increase the range of the instrument.

Volt Box

The range of the potentiometer can be increased by using a volt box to measure voltages above 1.6V. Figure 3 shows how a series of resistors can be used to drop the extra voltage and hence extend the range up to 75 volts. Using silver contacts and manganin resistors the error can be kept as low as 0.02%, but the design of the voltage divider presents a problem - resistors of low value are more stable and the galvanometer sensitivity is also greater - on the other hand, the current drain on the voltage source must be kept as low as possible, therefore a high resistance divider is called for.



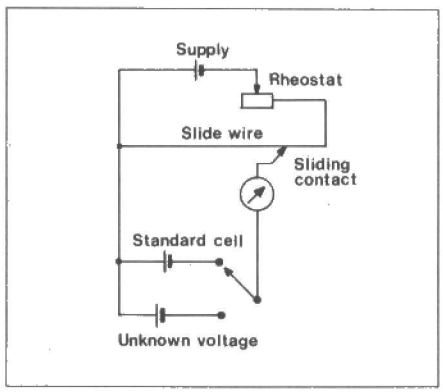


Figure 1. Basic Potentiometer.

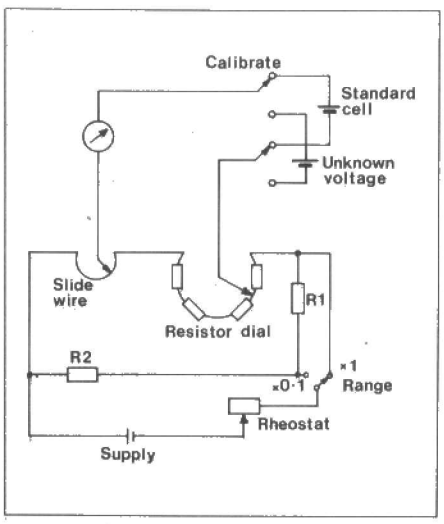


Figure 2. Potentiometer with ranges.

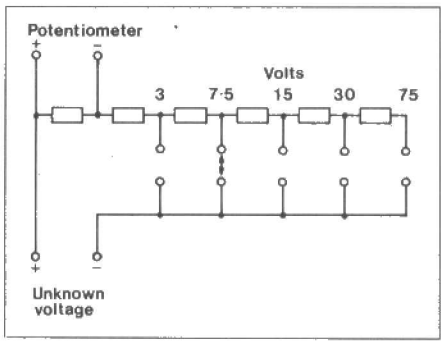


Figure 3. Volt Box.

Shunt Box

In order to measure current, a bypass or shunt is required to carry away the excess current that the meter cannot cope with. Figure 4 shows the arrangement which causes the current to divide into I_1 and I_2. These are in known proportions which are the proportions of the resistor chain. Only I_1 is allowed to excite the galvanometer and the current is read on the range selected.

As for the volt box, manganin resistors and silver contacts are used to obtain the 99.98% accuracy. With this kind of accuracy it is not surprising that this arrangement is used for calibrating DC ammeters and wattmeters.

Direct Current Meters

Measurements of currents and voltages may be divided into direct current (DC) and alternating current (AC). A DC voltage is one that is steady and not altering in polarity, see Figure 5a, whereas an AC voltage changes polarity, see Figure 5b. Any attempt to measure an AC voltage on a DC range will cause the meter needle to vibrate. This vibration will be visible if the rate of variation is low, e.g. 50Hz (mains frequency) or less.

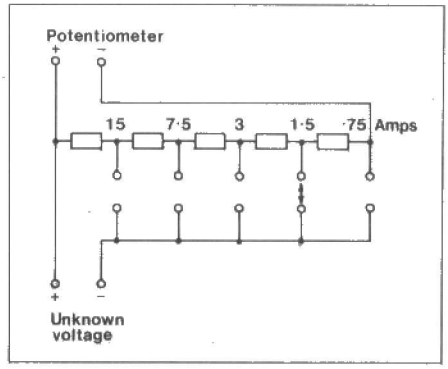


Figure 4. Shunt Box.

Galvanometers and PMMC's

A PMMC is a permanent magnet moving coil mechanism and a modern version of a suspension galvanometer. The construction of a suspension galvanometer is shown in Figure 6. Here, a coil of thin wire is suspended in the magnetic field of a permanent magnet. Current through the wire will create a magnetic field around it which reacts with the field of the magnet, causing the coil to rotate. It will rotate until the torque produced by the field equals the torsion of the wire. Therefore the deflection of the coil represents the current through the wire. A mirror attached to the coil deflects a beam of light along a scale to give the required current reading. The advantage of this method is that there is no needle to weigh down the coil.

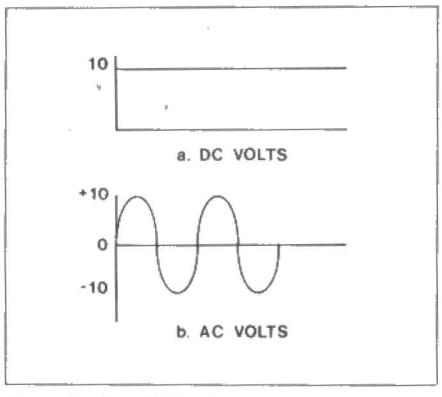


Figure 5. AC and DC voltages.

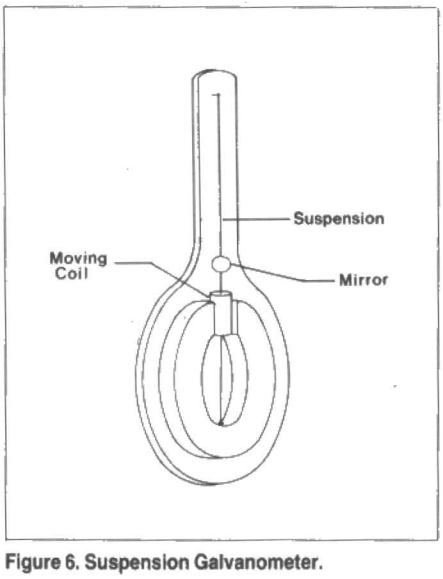


Figure 6. Suspension Galvanometer.

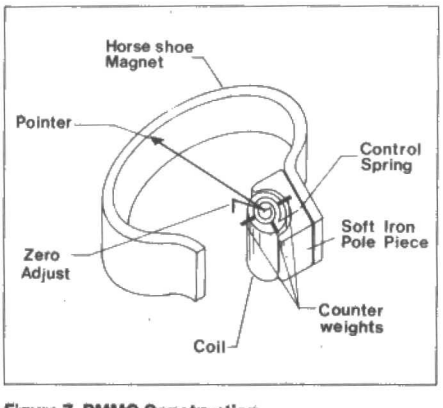


Figure 7. PMMC Construction.

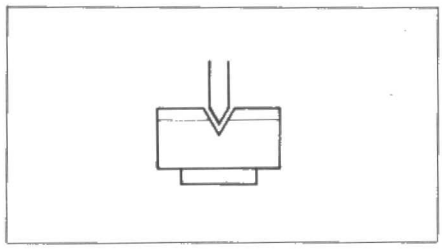


Figure 8. V groove mounting.

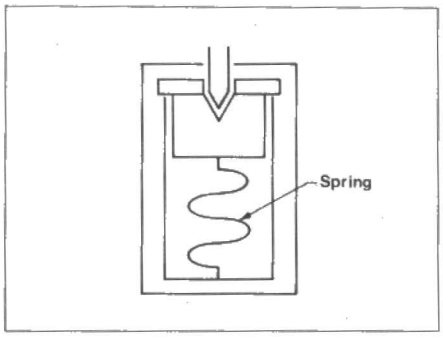


Figure 9. Incabloc mounting.

The above arrangement is delicate and unsuitable for use outside a laboratory. The construction of the more rugged PMMC is shown in Figure 7. This is also called the d'Arsonval movement, after the inventor. The d'Arsonval movement also has more refinements than the suspension galvanometer. For instance, the current carrying coil is not used to provide the opposing torque, instead two phosphor bronze springs of equal length are used. The thickness of the springs is controlled during manufacture to prevent the springs setting when stretched. The springs conduct the current to the coil which rotates between soft iron pole pieces mounted within the permanent magnet. The soft iron pole pieces ensure that there is a uniform magnetic field in the air gap. The back end of the needle has three counter weights which can be adjusted during assembly for various deflection positions. The whole rotating assembly is mounted in a V groove jewel bearing, see Figure 8, or the famous incabloc, shown in Figure 9, with spring back facilities, as used by watch manufacturers. The incabloc spring permits the assembly to move vertically if the instrument gets a sharp shock. The centre zero adjust is accessible from outside the glass cover.

The torque developed is given by:

$$T = B.A.I.N.$$

where T = torque in Newton metres
B = flux density in air gap (wb/m^2)
A = coil area (m^2)
I = current in amps
N = no. of turns on the coil.

Typical coil areas range from 0.5 to 2.5 square centimetres and flux densities from 0.15 to 0.5wb/m^2. A coil of 100 ohm resistance and carrying 1mA on full scale deflection would dissipate:

$$W = I^2R \text{ watts}$$
$$= (1x10^{-3})^2 100$$
$$= 100\mu W$$

Damping and Temperature Compensation

When a current is applied to the coil windings, the needle shoots forward until it is restrained by the springs and may bounce around before coming to rest. Similarly when the current is removed, the needle would fly back and bounce around the zero value before coming to rest.

This is similar to electric lifts travelling between floors of a building. Some overshoot the floor slightly before coming to rest. Others have a retarding force applied just before they reach the floor. The three cases of overdamping, underdamping and critical damping are shown by curves A, B and C respectively of Figure 10. We don't want the needle to come to its final reading in a slow sluggish manner, curve A. Neither do we have the time to watch it oscillate, curve B. A quick and effective reading is what we need, curve C. In practice, a slight overshoot is allowed, curve D. This allows some margin for wear and also indicates that the mechanism has not been damaged by mechanical shocks.

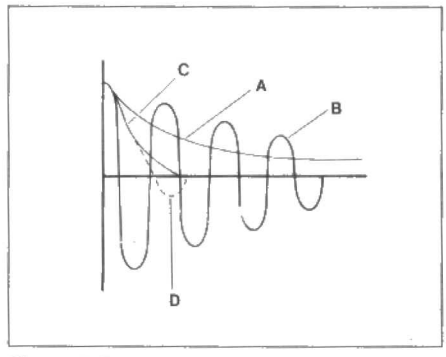


Figure 10. Damping.

Damping may be provided by mechanical or electromagnetic methods. Rotation of the coil provides friction with the air and with the bearings. In addition an aluminium vane may be connected to the coil. This vane moves in an air chamber and the amount of damping depends on the gap between the vane and the walls of the chamber. Electromagnetic damping is provided by connecting a resistor across the coil terminal. A back e.m.f. is induced in this circuit to provide damping. By trial, the value of resistor can be reduced, working one's way from oscillations down to critical damping. Another ploy is to wind the coil on an aluminium frame so that eddy currents are induced in the frame and hence an opposing torque is created.

An increase in temperature causes the spring tension to decrease and the meter reads high. The magnetic field strength also decreases with an increase in temperature while the coil resistance increases with a rise in temperature, causing the meter to read low. The net effect from these three factors (coil resistance, magnetic field strength, and spring tension) is to make the meter read low by about 0.2 per cent for every degree rise in temperature. A simple method of temperature compensation is to use a swamping resistor in series with the meter coil, Figure 11. The resistor is a combination of maganin and copper in ratios varying from 30:1 to 20:1. A

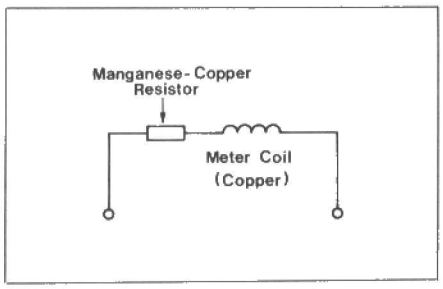


Figure 11. Temperature compensation.

more sophisticated compensation circuit is shown in Figure 12. The main disadvantage of swamping resistors is that the current through the meter is reduced and a higher applied voltage is required to maintain full scale deflection.

DC Volt Meters

To extend the meter reading beyond the basic range requires a resistor in series with the meter to drop the excess voltage, see Figure 13a. If the meter current is, say 1mA, for a full scale deflection of 1.5V and we want to measure 3V, then the extra 1.5V must be dropped across the series resistor. The series resistor, also called a multiplier resistor will have a value of:

$$\frac{1.5V}{1mA} = 1.5kohm$$

In this way several series resistors can be used, as Figure 13b, to give the required ranges. Alternatively, all the resistors can be in series and the range selector taps between the resistors, see Figure 13c. In the above simple calculation we have ignored the internal resitance (Rm) of the voltmeter. Let us assume in Figure 13c that Rm is 100 ohms and current for full scale deflection 1mA. Then, on the 2.5V range:

$$R_1 + R_m = \frac{2.5V}{1mA} = 2.5kohm$$

$$R_1 = 2,500 - 100 = 2,400 \text{ ohm}$$

Similarly, the total resistance required for measuring on the 5V range is:

$$\frac{5V}{1mA} = 5000 \text{ ohms}$$

But $R_1 + R_m = 2,500$ ohms

Hence $R_2 = 5000 - 2,500 = 2,500$ ohms

Depending on the instrument, the maximum range could be 250V or 500V or even 1000V. For higher ranges an external multiplier is used.

The sensitivity of a meter is quoted in ohms per volt and will be the same for all ranges. From the previous example, a full scale deflection of 1mA was obtained when 2.5V was applied across 2500 ohms or 5V across 5000 ohms. Therefore the sensitivity is 1000 ohms per volt. Sensitivity is an essential feature of any voltmeter since during measurement the instrument is across a circuit. If the circuit resistance is much lower than the meter sensitivity then an accurate reading is obtained. On the other hand if the circuit has a high resistance, the shunting effect of the meter loads the circuit down giving a lower voltage reading than the true reading.

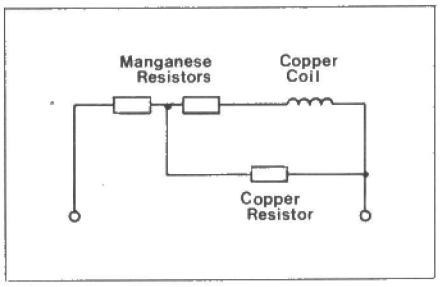


Figure 12. More sophisticated Temperature compensation.

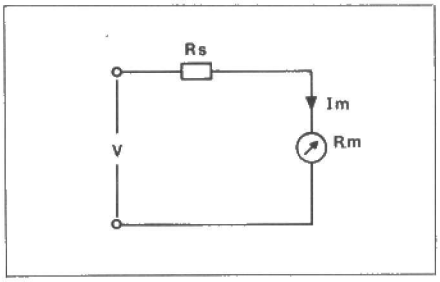


Figure 13a. Voltmeter with series resistor.

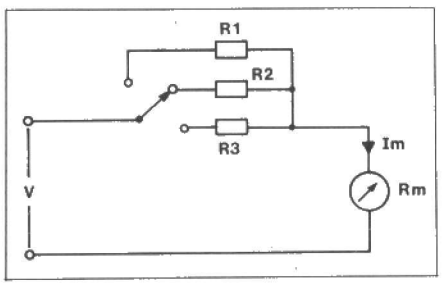


Figure 13b. Voltmeter with multirange.

To illustrate this, consider Figure 14. By calculation the voltage across R_1 is 50V, i.e. half the supply voltage is dropped across each resistor. One would expect to see this reading if one used a voltmeter. So let us use the above voltmeter with a sensitivity of 1000 ohms per volt. On the 50V range the total resistance presented by the meter would be 50 x 1000 = 50k. When this is across R_1 the effective resistance is only 250kohm. The 100V supply now sees R_1 = 25k and R_2 = 50k, therefore the meter will read:

$$\frac{25k}{75k} \times 100V = 33.3V$$

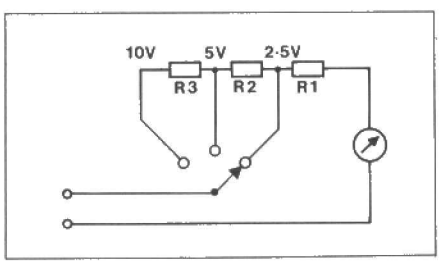


Figure 13c. Voltmeter with different multirange.

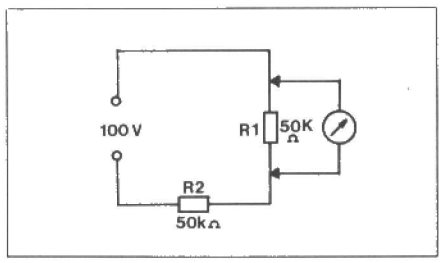


Figure 14. Meter sensitivity.

Let us now use a meter with a sensitivity of 20,000 ohm per volt. The total resistance on the 50V range is now 50 x 20,000 = 1Mohm. When this is across R_1 the effective resistance is 47.62k and the voltage reading:

$$\frac{47.62k}{97.62k} \times 100 = 48.78V$$

This latter reading of 48.78V is much closer to the true reading of 50V than 33.3V. This illustrates the need to be aware of an instrument's sensitivity. The other points worth noting are to start measuring on the highest possible range if the voltage is not known. The polarity of the meter terminals must also be observed to prevent damage to the instrument.

DC Current Meters

The maximum current through the meter must be restricted to that current which produces full scale deflection. In the voltmeter this was achieved by inserting a large enough series resistor to restrict the current. For measuring current a different arrangement is used. Here, the excess current bypasses the meter via a shunt, as Figure 15a. The value of this shunt resistor is calculated easily. From basic circuit theory, the voltage across two parallel arms is the same, i.e. the voltage across the meter is the same as that across the shunt. As before, if the meter accepts a full scale current of 1mA and has a coil resistance of 100 ohm then V = ImRm = 1mA x 100 = 0.1 volt.

Let us say we wish to measure 10mA, then the shunt must carry the current that the meter cannot, i.e. 10−1 = 9mA. The shunt resistance R_s will be:

$$\frac{0.1V}{9mA} = 11.1 \text{ ohm}$$

Several shunts can be used to give the required ranges, see Figure 15b. The special feature is the range selector switch which incorporates make before break in order not to leave the meter unprotected during range selection. Some large shunts (low resistance) would not be conventional resistors, but flat sheets of metal mounted outside the meter to permit heat dissipation.

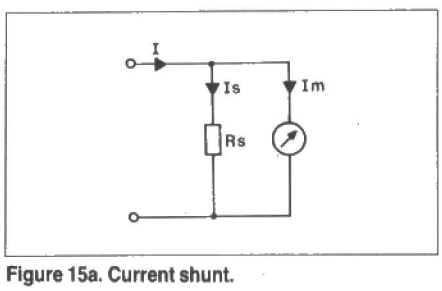


Figure 15a. Current shunt.

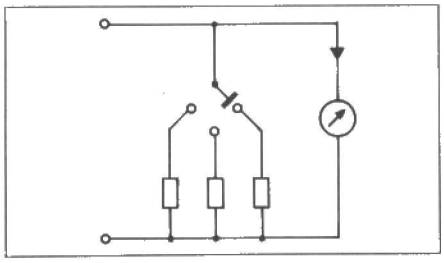


Figure 15b. Multirange shunt.

The Ayrton (universal) shunt does away with the need for a make before break switch, see Figure 15c. The highest range is always selected for measuring an unknown current and gradually reducing the resistance in Figure 15c until a reading is obtained on the meter.

The other safeguards of polarity, etc., that apply to voltmeters apply equally here to ammeters. Ranges of commercial current meters (ammeters) are from 20μA to 50A. External shunts can extend this to 500A. A current meter should *never* be connected across a voltage source without a limiting resistor. The values of the resistors in Figure 15c are calculated from simultaneous equations. There will be four simultaneous equations for the four unknowns, so let us consider the simpler case of Figure 16 with only two unknowns to give us 1A and 5A ranges. As before Rm = 100 ohm and Im = 1mA. Therefore the voltage is 0.1 volt. There is less resistance (only R_2) for the higher range (5A). So for the 1A range both R_1 and R_2 are in parallel with the meter and they have to pass the remaining current, i.e. 1A – 1mA = 999mA.

$$R_1 + R_2 = \frac{0.1}{999mA} = 0.1001 \text{ ohm}$$

In the 5A range, only R_2 is in parallel with Rm and R_1 so with 1mA going through these, 4,999mA must pass through R_2:

$$R_2 = \frac{1mA \times (R_1 + Rm)}{4,999mA} = \frac{R_1 + 100}{4,999}$$

Substituting this value of R_2 in the above equation gives:

$$R_1 + \frac{R_1 + 100}{4,999} = 0.1001$$

$$4999R_1 + 100 = 500.39$$
$$5000R_1 = 400.39$$
$$R_1 = 0.08007 \text{ ohm}$$

Substituting this value for R_1 in $R_1 + R_2 = 0.1001$ gives $R_2 = 0.02003$ ohm. This calculation shows how small the shunt resistances are and for higher ranges, these shunts are even smaller.

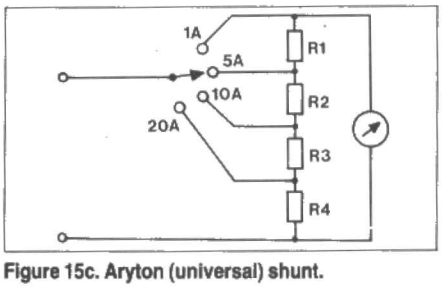


Figure 15c. Aryton (universal) shunt.

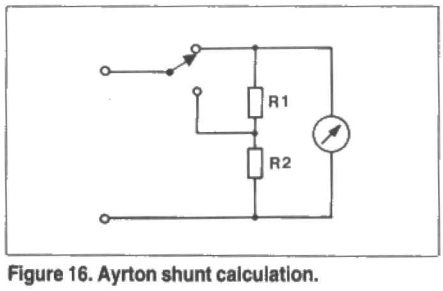


Figure 16. Ayrton shunt calculation.

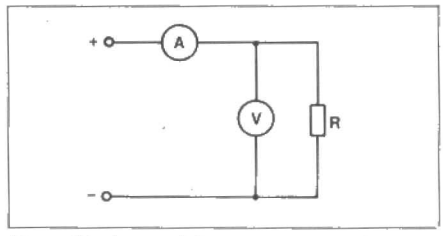


Figure 17a. Low resistance measurement.

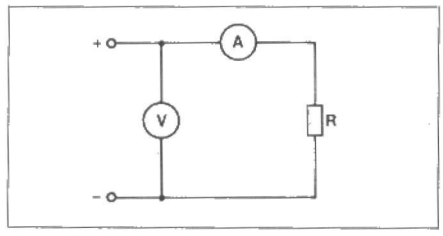


Figure 17b. High resistance measurement.

Ohmmeters and Resistance Measurements

Resistance measurement does not require an ohmmeter since an unknown resistance can be calculated from values of voltage and current in the circuit, see Figure 17. This assumes that the voltmeter and ammeter do not load the circuit in any way. In practice the ammeter will drop some voltage across it and the voltmeter will draw some current. Therefore Figure 17a is the best arrangement for measuring low resistance and Figure 17b for high resistance. It is easy enough to design an ohmmeter from a basic PMMC movement. The ohmmeter may be of the shunt or series type, Figure 18a and 18b respectively. In the shunt ohmmeter, the power supply, to unknown resistor and meter are in parallel with each other. It is not used very often outside laboratories but it is useful for measuring low values of resistance. R_1 compensates for battery ageing and is controlled externally for resetting the zero reading. That is, with terminals A and B shorted, the current flows through the short and the meter needle does not move from its rest position on the left. With A and B open circuit, i.e. an infinite resistance the meter reads full scale since all the current flows through the meter. Known values of resistance can be connected to calibrate

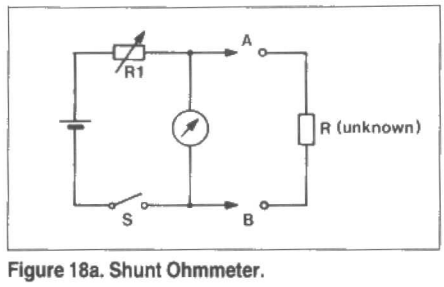


Figure 18a. Shunt Ohmmeter.

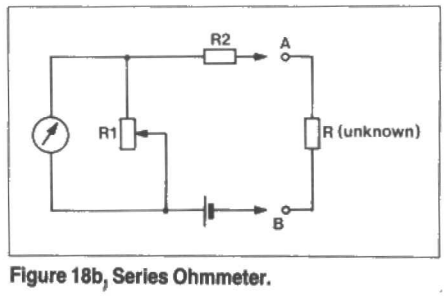


Figure 18b, Series Ohmmeter.

intermediate positions on the scale. The switch S is required to disconnect the meter from the battery when not in use.

In the series ohmmeter, the power supply, meter and unknown resistor are all in series. This means that if terminals A and B shorted, the meter reads full scale, i.e. zero reading is on the right hand side. Similarly if A and B are open circuited, infinite resistance, the needle remains at rest on the left. Once again intermediate values are marked on the scale by connecting known values of resistor to A and B. R_2 is a series limiting resistor to prevent damage to the meter. The set zero resistor R_1 to compensate for battery ageing is now in parallel with the battery, i.e. it drains some of the current from a new battery. When the battery voltage drops due to age, R_1 will have to be increased to reduce the drain.

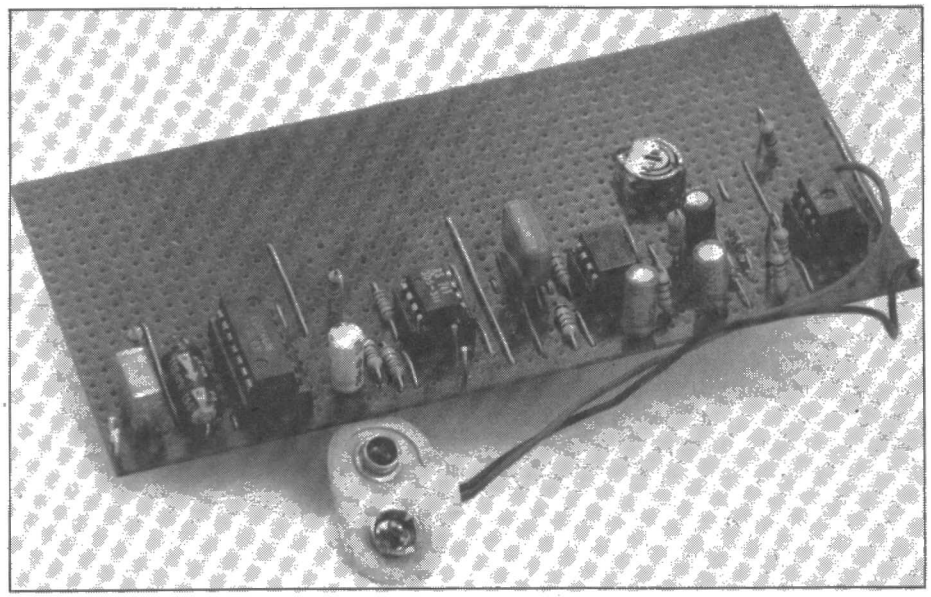
Ohmmeters are of limited accuracy but sufficient for most fault finding exercises. If accurate measurements are required then a bridge method must be employed.

Multimeters

All the above principles are incorporated in a multimeter used for measuring voltage, current and resistance. The cheaper models have fewer ranges while the larger, more expensive models have more ranges. The resistance measurement usually comes in two or three ranges X1, X10, X100. A 1.5V cell serves the lower ranges but a larger cell (say 15V) is used on the higher range. Voltage ranges from 0 to 1,000V are typical, usually in steps 0-2.5V, 0-10V, 0-250V. Current ranges are typically 0-1mA, 0-10mA, 0-100mA, and up to 10A.

The circuits discussed above are DC circuits but of course, the instrument must also be capable of measuring AC currents and voltages and such circuits will be the subject of the next part of this series. Therefore all AC-DC instruments incorporate a selector to switch between the AC and DC measuring circuits. Instruments with a pointer movement are called analogue displays and were of the earlier type. Today, more and more instruments giving a digital readout, i.e. in numbers, are making their appearance. The principles of these instruments will be described later in the series.

Noise Gate



Although advances in electronics have produced ever increasing levels of fidelity with steady improvements in signal to noise ratios, excessive background noise can still be a problem sometimes. A simple way of combatting noise is to use a device known as a noise gate. This is a form of electronic switch, and it permits the signal to pass if it has an amplitude above a certain threshold level, or else the signal is blocked if it is less than this level. The idea is to set the threshold level so that it is just above the amplitude of the noise, and therefore cuts the signal path when only the noise is present.

When a proper signal is present, it raises the overall signal level above the switching threshold, and enables the signal to pass. Thus the circuit cuts out the background noise during gaps in the main signal. This is admittedly only a fairly crude form of noise reduction, but under many circumstances, it can be quite effective. It would best find applications in communications or effect units.

The circuit is built around IC2, a quad transition gate CMOS 4066BE device, containing four separate bi-directional on/off switches. IC1 acts as an inverting amplifier with approximately unity voltage gain, but with IC2 switched to the 'on' state, the resistance of the switch forms an attenuator in conjunction with R2. As the resistance of the switch is only around 200Ω, this gives an attenuation of well over 40dB, and this effectively cuts off low level signals completely. In some circumstances, a much lower level of attenuation might be preferred, and this can be achieved by adding a resistor in series with IC2.

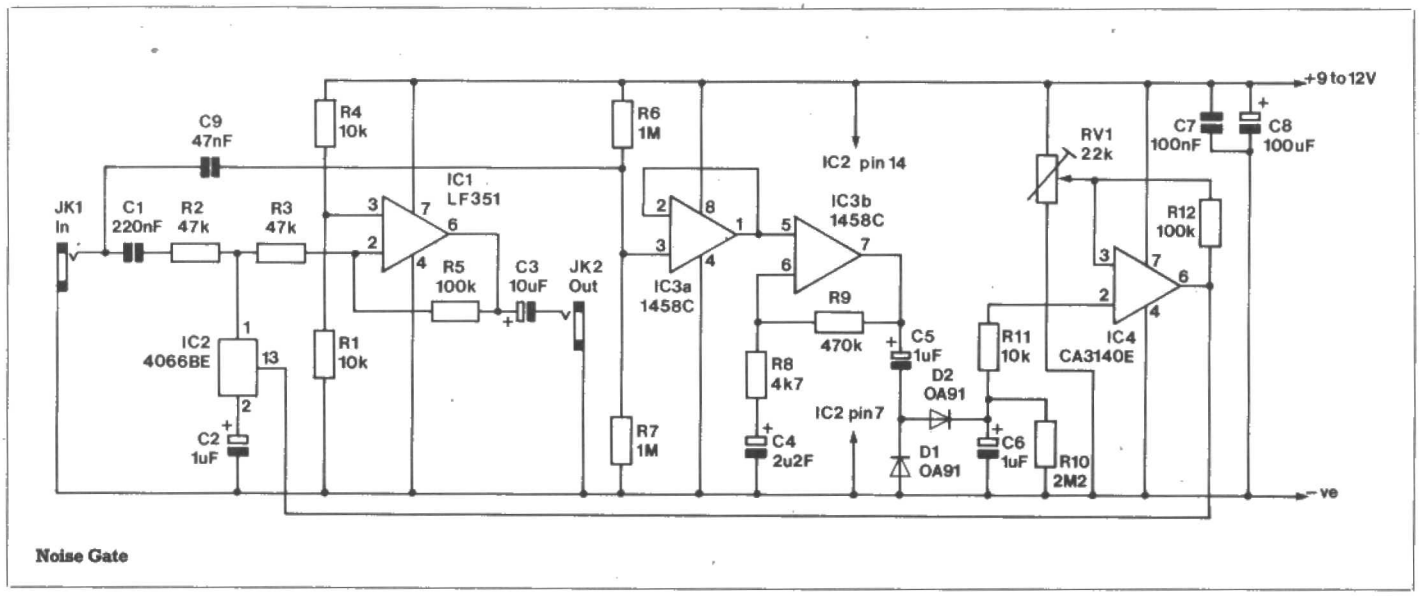
When any gate of IC2 is switched off it has an extremely high resistance, and it does not introduce any significant distortion, or have any other adverse effects on the circuit.

IC2 requires standard CMOS input levels, with the four control inputs going 'high' to set the gates in the 'on' state. Only one of these gates is used for one noise gate circuit, where the control voltage is provided by a Schmitt Trigger to provide a proper switching waveform. This is generated by first tapping off some of the input signal via the buffer stage based on IC3a. This has a high input impedance of about 500kΩ so that it does not significantly reduce the input impedance of the circuit (which is about 80kΩ overall). The output of IC3a is amplified by IC3b, and this stage is a straightforward non-inverting amplifier, having a voltage gain of approximately 40dB (V x 100) as defined by negative feedback network R8–R9.

The amplified signal is rectified by D1 and D2, and then smoothed by C6. IC4 operates as the inverting Schmitt Trigger with positive feedback (and a small amount of hysteresis) provided by R12. RV1 controls the threshold levels of the trigger circuit, and effectively operates as the sensitivity control in this application. The output of the smoothing circuit (from C6) is coupled to the input of the Schmitt Trigger via a protection resistor, R11.

If the input signal is suitably strong, then the DC voltage to the Schmitt Trigger circuit will be made adequate to set the output to the low state, thus switching off IC2. With an inadequate input signal level the output of IC4 goes high and switches on IC2, giving the required gating action. The attack time of the circuit is quite short at just a few milliseconds, but the value of R10 has been made quite high so that the smoothing circuit has a long decay time of several seconds. The decay time can be shortened if required simply by using a lower value for R10, or a variable decay time can be obtained by replacing R10 with a 2M2Ω linear potentiometer.

For stereo use two noise gate circuits are required, with each channel being processed separately. Note that one of the unused switches in IC2 can be utilised in the second channel. When constructing the unit bear in mind that IC2 and IC4 are both MOS devices which require the usual antistatic handling precautions. Also, D1 and D2 are germanium diodes which are vulnerable to heat damage, and due care should be exercised when soldering these into place. It may be a good idea to leave a length of lead at least ¼in. from the glass body, and sleeved with insulation stripped from connecting wire, assuming that the diode would otherwise use pcb holes further apart than ½ inch.

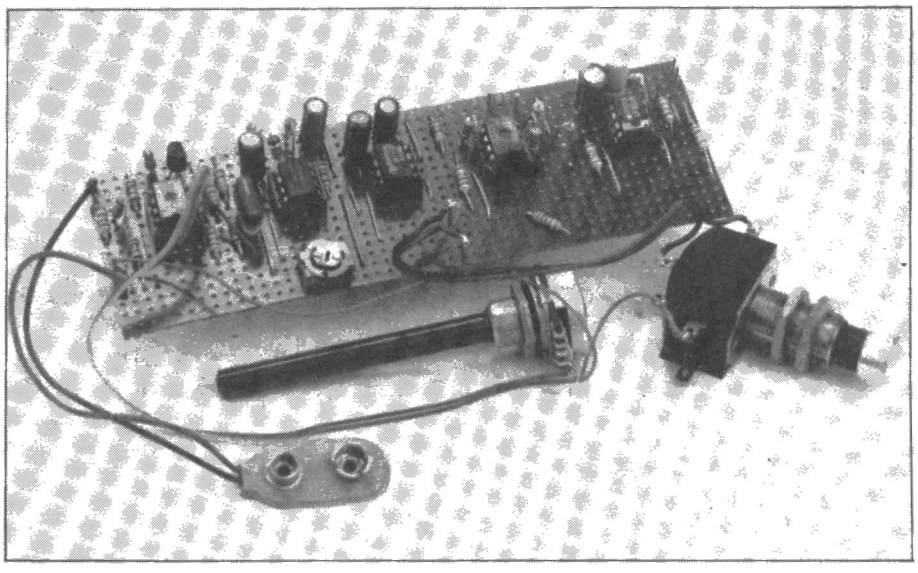


Noise Gate

Envelope Tremolo

Tremolo is perhaps the oldest of electronic musical effects, and works on the principle of simply amplitude modulating the input signal, or varying its volume in other words. With the conventional method the modulation is provided by a low frequency sinewave or triangular waveform generator, with an adjustable frequency range of around 0.5 to 10Hz. This gives quite a useful effect, but one which sounds rather mechanical due to the lack of any change in modulation rate during the course of each note.

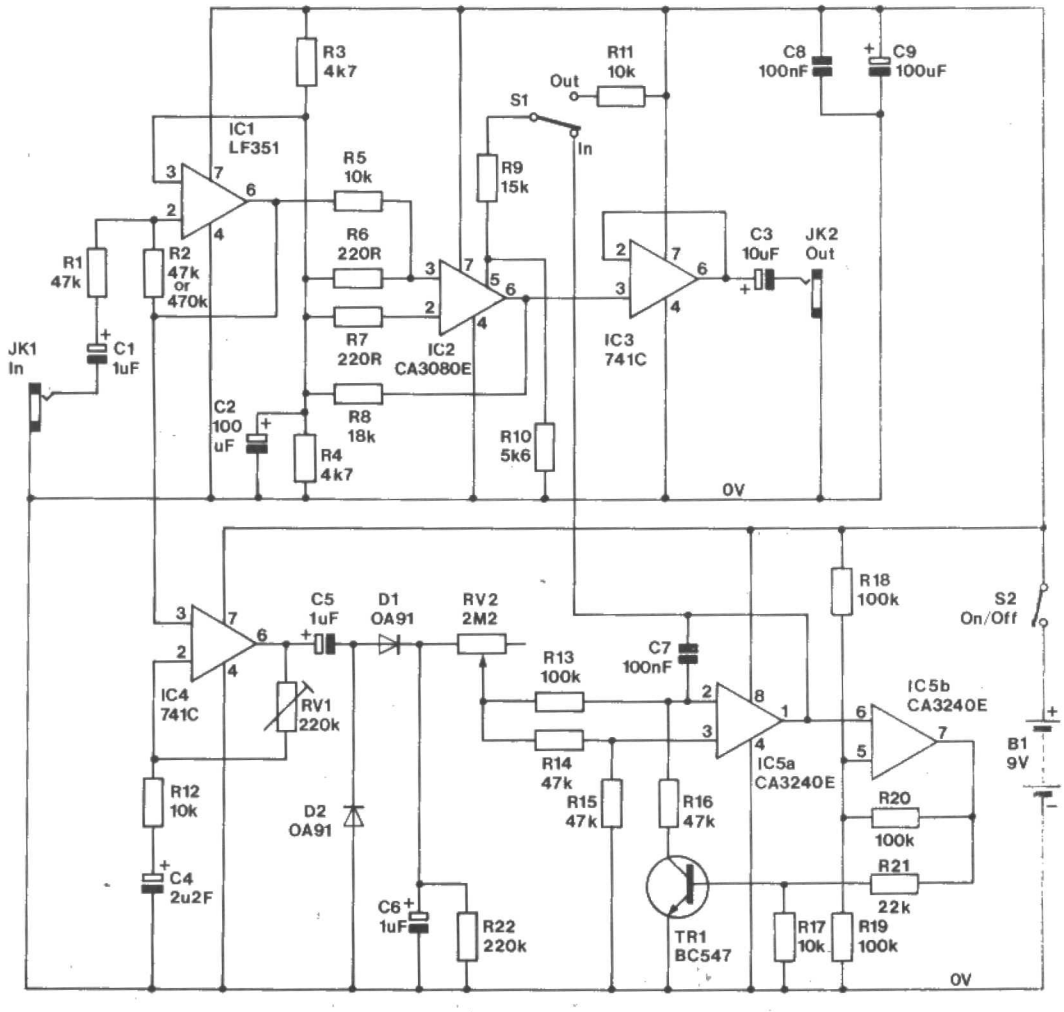
A much more interesting effect can be obtained by having the modulation frequency related to the envelope of the input signal, and it is this approach that has been adopted for this tremolo unit. The modulation frequency starts at a relatively high figure when the guitar note is first struck and the volume is high, and it gradually reduces as the volume of the note decays.



IC1 acts as an input buffer and amplifier stage. If the unit is to be used with a high output guitar, or some other electronic instrument such as a monophonic synthesiser, R2 should have a value of 47k so that IC1 acts as a unity voltage gain buffer stage. For use with a low output guitar, R2 should be increased to 470k so that IC1 provides a voltage gain of about 11 times, and boosts the output of the guitar to an adequate level to drive this unit properly.

The voltage controlled amplifier is based on IC2 and IC3. IC2 is a transconductance operational amplifier which is connected in a standard VCA configuration with IC3 acting as the output buffer stage. The bias voltage fed to R9 controls the gain of the VCA, and with S1 in the 'OUT' position a fixed bias is provided so that the input signal passes through the unit without any modulation being introduced: When set to the 'IN' position, S1 connects the control input of the VCA to the output of a voltage controlled oscillator.

The VCO is built around IC5, which is used in a circuit that is very similar to an ordinary triangular/squarewave oscillator of the type which is based on a Miller Integrator and a Schmitt Trigger. In fact, IC5a operates as an integrator,



Envelope Tremolo

and IC5b acts as a trigger circuit, but the standard configuraiton has been modified somewhat to provide voltage controlled operation. The control voltage is applied to the junction of R13 and R14. It is, of course, the triangular output from IC5a that is used as the modulation signal, and not the squarewave from IC5b.

The control signal is generated by first amplifying the output from IC1 using IC4 as a non-inverting amplifier. The output from IC4 is then rectified and smoothed by D1, D2 and D6 to give a DC control voltage which is roughly proportional to the amplitude of the input signal. RV2 enables the maximum modulation frequency to be controlled.

The finished unit only requires one adjustment before it is ready for use, and this is to set RV1 for a suitable voltage gain through IC4. This is really a matter of trial and error to find a setting that gives good results. If set too low in value, the voltage gain of the circuit will be inadequate, and only very low modulation frequencies will be obtained even with RV2 set at minimum resistance (which corresponds with maximum modulation frequency range). Setting RV1 at an excessive resistance will result in the modulation frequency remaining virtually constant at a high figure until the note has almost fully decayed. When RV1 is set correctly, a fairly high maximum modulation frequency should be attainable, but the frequency should start to reduce downwards soon after the beginning of each note.

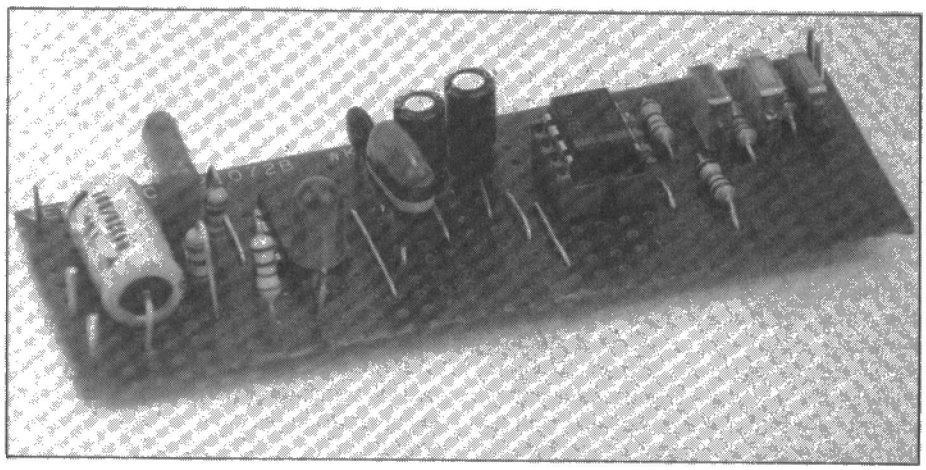
RTTY Tone Decoder

A sophisticated RTTY encoder/decoder was described in issue 12 of this magazine, and this unit is capable of excellent results. However, if reception of RTTY is all that is required, it is possible to use a much more simple and less expensive circuit, such as the simple RTTY decoder described here. There are two basic types of RTTY tone decoder, the filter/level detector type, and circuits based on phase locked loops. Although regarded by many as old fashioned, the filter/level detector designs (including the Maplin TU1000) offer excellent performance and are superior to many phase-locked-loop designs. The advantage of phase locked loop decoders, and the reason for using one in this case, is that they offer extreme simplicity.

The phase locked loop device used in this circuit is the NE567 (IC2). This offers good sensitivity, and it will operate with inputs of as little as 20 millivolts RMS. There should therefore, be no problem in operating the unit from an 'ext. speaker', 'phone', or 'line' output of the receiver. R7 and C9 are the timing components for the VCO, and these give a centre frequency of about 1.6kHz. C8 is the filter capacitor in the lowpass filter. The NE567 has an unusual feature in the form of a syncronous detector. Basically this is an electronic switch which is operated by the output of the VCO, and it connects the input signal through to filter capacitor C7 during positive VCO half cycles. If the VCO is locked onto the input signal, the switch half-wave rectifies the input signal and C7 takes up a strong positive charge.

If it is not, the output from the switch is randomly phased and gives no significant charge on C7. A voltage detector determines whether or not there is a strong charge on C7, and if there is, it switches on an npn output transistor. The latter has R5 plus LED indicator D1 as its collector load.

In order to provide RTTY decoding, the receiver is simply tuned so that one of the audio output tones is within the capture range of IC2. When this tone is present, the output transistor switches on and the output goes low. When this tone is absent, the output transistor cuts off and the output goes high, giving the required tone decoding action. No 'invert' facility is included in the circuit,



since the phase of the output signal can be changed simply by tuning the receiver so that the unit decodes the higher tone rather than the lower one, or vice versa.

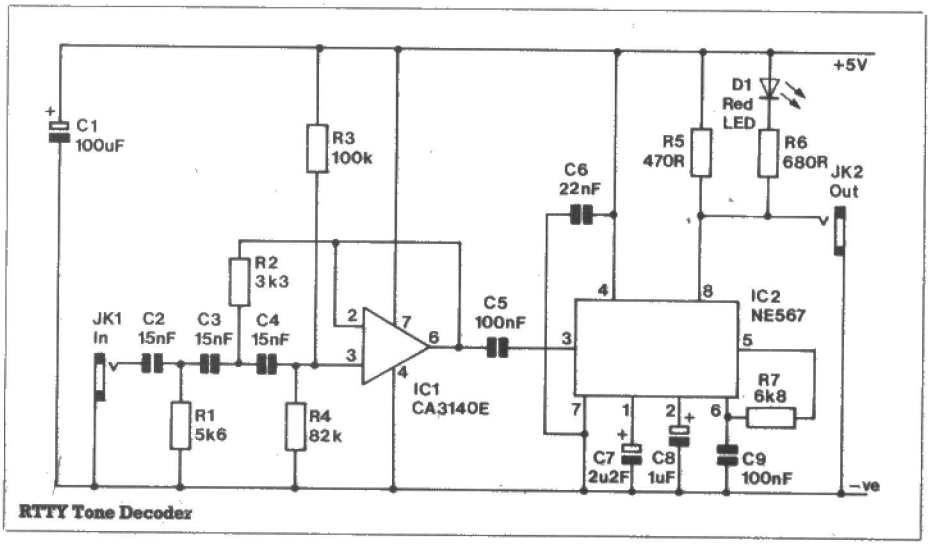
The output from JK2 is at standard 5 volt logic levels, and it can be decoded by a UART. It will also drive most RS232C inputs despite the fact that it does not provide full RS232C levels. The TU1000 article gives more information on Baudot codes and decoding software.

IC1 is used as the basis of a third order highpass filter having a cut-off frequency of about 1.4kHz. Together with the IF filtering of the receiver, this restricts the passband to that which is more suitable for RTTY reception.

The unit is very easy to use, and it is just a matter of tuning the receiver to lock IC2 onto one of the tones. D1 acts as a simple tuning indicator as it switches on when lock is achieved, and it flashes on and off in sympathy with the tone modulations. If just random characters appear on the screen, and assuming the baud rate you are using is correct, the problem is probably that the output signal is of the wrong polarity, and you should tune to the other tone to correct this.

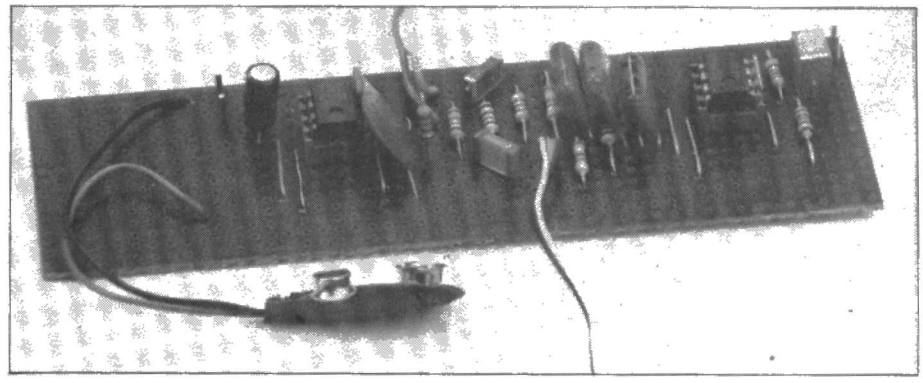
The circuit requires a single 5 volt supply and has a current consumption of only about 10 milliamps. The computer used to decode the serial output signal should be able to supply this.

It should be emphasised that this very simple circuit is probably best used for experimentation prior to purchasing a full blown system, such as the TU1000, and the circuit works best under ideal conditions.



RTTY Tone Decoder

Scratch and Rumble Filter



With the advent of the compact disk player, problems with scratched and warped records are now something that can be avoided, but this is of little comfort to someone who has a large library of ordinary microgroove records, most of which will never be released as compact disks, and the cost of the changeover would be prohibitively expensive for most people anyway. Records, warps-and-all, seem likely to be with us for many years to come.

The three most common problems with old recordings are surface noise, deep scratches, and warps. The surface noise is caused by minute particles of dust in the grooves, plus wear on the groove walls, in the case of a well used recording. The result is the familiar background 'crackling' sound.

Record warps produce low frequency signals with much of the output in the sub-audio band. These signals, although not directly audible, can produce audible effects by shifting the amplifier's signal path biasing levels, causing movement of the loudspeaker diaphragm in sympathy, away from its central rest position. This can result in signal clipping distortion in the amplifier as the signal path 'runs out of headroom', and severe distortion from the speaker as it is forced to operate from an off-centre position.

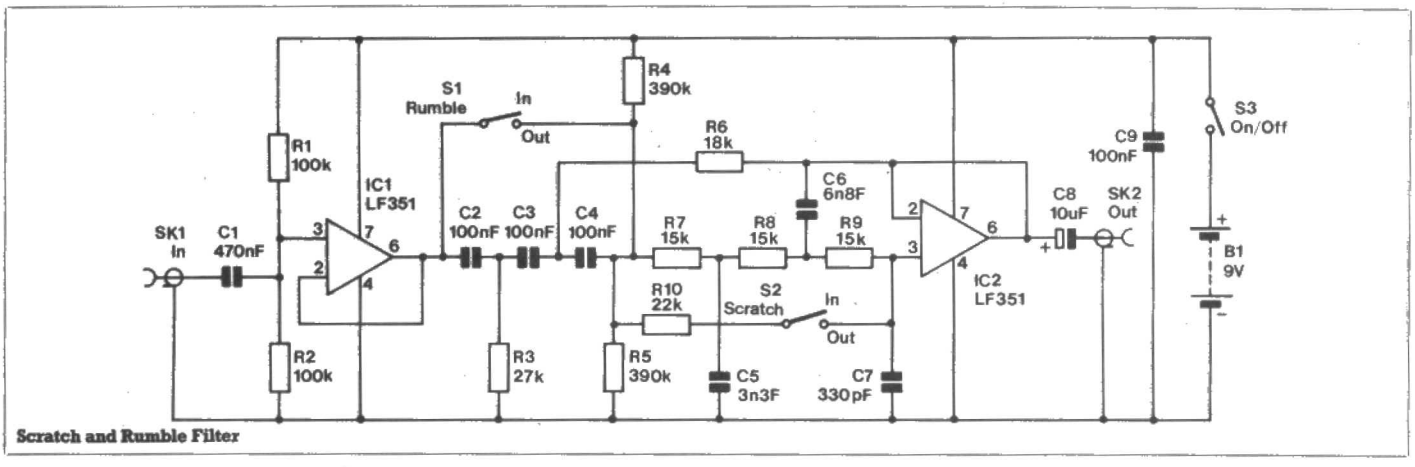
Both problems can be effectively countered by filtering, although this is at the expense of audio quality in that some of the wanted signals are removed together with the noise. Overall, most people prefer the slight loss of quality to enduring the problems of record scratches and warps. The circuit shown here is an active filter which provides lowpass filtering to combat scratches, and highpass filtering to combat record warps. It is only fair to point out that deep scratches can not be effectively countered by simple filtering since the signals generated by these cover a large part of the audio spectrum.

IC1 is an input buffer stage which provides an input impedance of about 50k. IC2 is another unity gain buffer stage, and it is the active element for both filters. The highpass filtering is provided by C2 to C4, and R3 to R6. This is a third order (that is 18dB per octave) type with a cut-off frequency at about 40Hz. The cut-off frequency is easily changed if desired, and it is just a matter of altering the value of C2 to C4. Changes in value alter the cut-off frequency by an inversely proportional amount. S1 can be used to switch out the rumble filtering.

The lowpass filtering is provided by R7 to R9, and C5 to C7. Again, this is a third order filter. The cut-off frequency is at about 6.5kHz but if required this can be altered by changing the value of resistors R7 to R9. Increasing or decreasing these values give an inversely proportional change in frequency.

When S2 is closed the cut-off frequency is pushed above the upper limit of the audio band, effectively removing the scratch filtering. Although R10 may seem to serve no useful purpose, without this resistor the circuit would break into oscillation when S2 was closed.

Although the circuit shown here is only for a single monophonic channel, it is merely necessary to build two units for stereo operation, one for each channel. The battery and on/off switch can be common to both channels, and S1 and S2 would be double pole types with one pole used in each channel. Ideally, the unit should be connected between the preamplifier and power amplifier stages so that it is operating with a fairly high input signal level. This gives a much better signal to noise ratio than that obtained if the unit is added between the cartridge and the input of the pre-amplifier.



Scratch and Rumble Filter

SCRATCH AND RUMBLE FILTER PARTS LIST

RESISTORS: All 0.6W 1% Metal Film

R1,2	100k	2	(M100K)
R3	27k	1	(M27K)
R4,5	390k	1	(M390K)
R6	18k	1	(M18K)
R7-9	15k	3	(M15K)
R10	22k	1	(M22K)

CAPACITORS

C1	470nF poly-layer	1	(WW49D)
C2-4	100nF poly-layer	3	(WW41U)
C5	3n3F poly-layer	1	(WW25C)
C6	6n8F poly-layer	1	(WW27E)
C7	330pF ceramic	1	(WX62S)
C8	10μF 50V PC Electrolytic	1	(FF04E)
C9	100nF ceramic	1	(BX03D)

SEMICONDUCTORS

IC1,2	LF351	2	(WQ30H)

MISCELLANEOUS

S1—3	SPST ultra-min toggle	3	(FH97F)
SK1,2	Phono socket	2	(YW06G)
B1	Battery PP3 9 volt	1	(FK58N)
	Battery clip	1	(HF28F)
	8-pin DIL socket	2	(BL17T)

BASS FUZZ PARTS LIST

RESISTORS: All 0.6W 1% Metal Film			
R1,2	22k	2	(M22K)
R3	4k7	1	(M4K7)
R4	100k	1	(M100K)
R5-8	10k	4	(M10K)
RV1	220k hor sub-min preset	1	(WR62S)
RV2	4k7 hor sub-min preset	1	(WR57M)
CAPACITORS			
C1,7	10μF 50V PC Electrolytic	2	(FF04E)
C2	220nF poly-layer	1	(WW45Y)
C3	4n7F poly-layer	1	(WW26D)
C4	6n8F poly-layer	1	(WW39N)
C5	22nF poly-layer	1	(WW33L)
C6	1nF poly-layer	1	(WW22Y)
C8	100μF 10V PC Electrolytic	1	(FF10L)
SEMICONDUCTORS			
IC1,2	LF351	2	(WQ30H)
D1,2	1N4148	2	(QL80B)
MISCELLANEOUS			
JK1,2	¼in. standard jack socket	2	(HF90X)
S1	SPST ultra min toggle	1	(FH97F)
B1	Battery PP3 9 volt	1	(FK58N)
	Battery clip	1	(HF28F)
	8-pin DIL socket	2	(BL17T)

VOICE-OVER UNIT PARTS LIST

RESISTORS: All 0.6W 1% Metal Film			
R1,2,17	4k7	3	(M4K7)
R3	1M	1	(M1M)
R4,5	2k2	2	(M2K2)
R7,9	22k	2	(M22K)
R14	10k	1	(M10K)
R15	2M2	1	(M2M2)
R6,8,10-13,16	470k	7	(M470K)
R18,19	100k	2	(M100K)
RV1	47k sub-min hor preset	1	(WR60Q)
CAPACITORS			
C1,2	470nF poly-layer	2	(WW49D)
C3,10	100μF 10V PC Electrolytic	2	(FF10L)
C4,5	10μF 50V PC Electrolytic	2	(FF04E)
C6,7	1μF 100V PC Electrolytic	2	(FF01B)
C8	4μ7F 63V PC Electrolytic	1	(FF03D)
C9	330nF poly-layer	1	(WW47B)
SEMICONDUCTORS			
IC1	4007UBE	1	(QX04E)
IC2	LF353	1	(WQ31J)
IC3	CA3140E	1	(QH29G)
IC4	1458C	1	(QH46A)
D1	1N4148	1	(QL80B)
D2,3	0A91	2	(QH72P)
MISCELLANEOUS			
SK1-5	Phono socket	5	(YW06G)
	8-pin DIL socket	3	(BL17T)
	14-pin DIL socket	1	(BL18U)

NOISE GATE PARTS LIST

RESISTORS: All 0.6W 1% Metal Film			
R1,4,11	10k	3	(M10K)
R2,3	47k	2	(M47K)
R5,12	100k	2	(M100K)
R6,7	1M	2	(M1M)
R8	4k7	1	(M4K7)
R9	470k	1	(M470K)
R10	2M2	1	(M2M2)
RV1	22k hor sub-min preset	1	(WR59P)
CAPACITORS			
C1	220nF poly-layer	1	(WW45Y)
C2,5,6	1μF 100V PC Electrolytic	2	(FF01B)
C3	10μF 50V PC Electrolytic	1	(FF04E)
C4	2μ2F 100V PC Electrolytic	1	(FF02C)
C7	100nF ceramic	1	(BX03D)
C8	100μF 25V PC Electrolytic	1	(FF11M)
C9	47nF poly-layer	1	(WW37S)
SEMICONDUCTORS			
IC1	LF351	1	(WQ30H)
IC2	4066BE	1	(QX23A)
IC3	1458C	1	(QH46A)
IC4	CA3140E	1	(QH29G)
D1,2	0A91	2	(QH72P)
MISCELLANEOUS			
JK1,2	¼in. standard jack socket	2	(HF90X)
	8-pin DIL socket	3	(BL17T)
	14-pin DIL socket	1	(BL18U)

ENVELOPE TREMOLO PARTS LIST

RESISTORS: All 0.6W 1% Metal Film			
R1,14-16	47k	4	(M47K)
R2	See Text		
R3,4	4k7	2	(M4K7)
R5,11,12,17	10k	4	(M10K)
R6,7	220Ω	2	(M220R)
R8	18k	1	(M18K)
R9	15k	1	(M15K)
R10	5k6	1	(M5K6)
R13,18,19,20	100k	4	(M100K)
R21	22k	1	(M22K)
R22	220k	1	(M220K)
RV1	220k sub-min hor preset	1	(WR62S)
RV2	2M2 lin pot	1	(FW09K)
CAPACITORS			
C1,5,6	1μF 100V PC Electrolytic	3	(FF01B)
C2,9	100μF 10V PC Electrolytic	2	(FF10L)
C3	10μF 50V PC Electrolytic	1	(FF04E)
C4	2μ2F 100V PC Electrolytic	1	(FF02C)
C7	100nF poly-layer	1	(WW41U)
C8	100nF ceramic	1	(BX03D)
SEMICONDUCTORS			
IC1	LF351	1	(WQ30H)
IC2	CA3080E	1	(YH58N)
IC3,4	μA741C	2	(QL22Y)
IC5	CA3240E	1	(WQ21X)
TR1	BC547	1	(QQ14Q)
D1,2	0A91	2	(QH72P)
MISCELLANEOUS			
JK1,2	¼in. Standard jack	2	(HF90X)
S1	Press Toe Switch SPDT	1	(FH92A)
S2	SPST ultra-min toggle	1	(FH97F)
B1	Battery PP3 9 volt	1	(FK58N)
	Battery clip	1	(HF28F)
	8-pin DIL socket	5	(BL17T)

RTTY DECODER PARTS LIST

RESISTORS: All 0.6W 1% Metal Film			
R1	5k6	1	(M5K6)
R2	3k3	1	(M3K3)
R3	100k	1	(M100K)
R4	82k	1	(M82K)
R5	470Ω	1	(M470R)
R6	680Ω	1	(M680R)
R7	6k8	1	(M6K8)
CAPACITORS			
C1	100μF 10V PC Electrolytic	1	(FF10L)
C2-4	15nF poly-layer	3	(WW31J)
C5,9	100nF poly-layer	2	(WW41U)
C6	22nF poly-layer	1	(WW33L)
C7	2μ2F 100V PC Electrolytic	1	(FF02C)
C8	1μF 100V PC Electrolytic	1	(FF01B)
SEMICONDUCTORS			
IC1	CA3140E	1	(QH29G)
IC2	NE567	1	(QH69A)
D1	LED Red	1	(WL27E)
MISCELLANEOUS			
JK1,2	3.5mm jack socket	2	(HF82D)
	8-pin DIL socket	2	(BL17T)

The Story of Radio

by Graham Dixey
C.Eng., M.I.E.R.E.

Part Five- The 1920's

After the Great War (the war to end all wars, so it was thought), little progress was made with wireless in the services for many years. However, there were those who appreciated that, what could be used successfully at sea, could also find application in the air. When Alcock and Brown made the first non-stop Transatlantic flight in 1919, they carried wireless in their Vickers Vimy biplane in order to maintain some contact with the outside world. It is hardly their fault that, shortly after take-off, the propeller of their wind-driven generator broke, leaving them in isolation for the whole of the flight. More successfully, the first regular passenger air service between London and Paris relied upon wireless for communication and navigation, and ground stations were built at Croydon, Lympne and Le Bourget in 1920. The first flight of a civil airliner equipped with wireless happened on the 4th March 1920, when a Handley-Page, registration number G-EALX, equipped with a Marconi transmitter, with 100W DC input and a 5-valve receiver, flew the route from Cricklewood to Paris. Navigational assistance through fog en route was given from Croydon station.

The Roaring Twenties

Before the middle of the 1920's regulations were issued that required certain classes of aircraft to carry wireless,

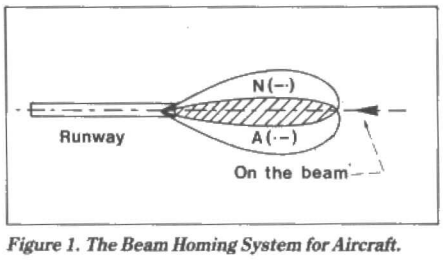


Figure 1. The Beam Homing System for Aircraft.

as had already been done with ships at sea. Radio aids to navigation became essential and direction finding equipment (d.f.) carried in the aeroplane was developed. Also developed was the 'beam homing system', in which the Morse code for 'A' (. -) was transmitted on a beam alongside another carrying the Morse letter 'N' (- .), on a given bearing. An aircraft 'on the beam' heard only the dash (where the beams overlapped) but heard either A or N separately if it deviated to one side or the other (Figure 1). Thus, wireless was used in aircraft for the peaceful purpose of conveying people safely from A to B, a far cry from when it was recently used to call down explosive shells on the heads of the enemy.

Meanwhile at sea, valves gradually came into use more and more, though the spark transmitter remained in service for quite a long time, some examples still being seen as late as the beginning of the Second World War. Another innovation was ship-to-shore radio telephony (i.e. the transmission of speech as opposed to code), while the marine sphere saw some lifeboats being

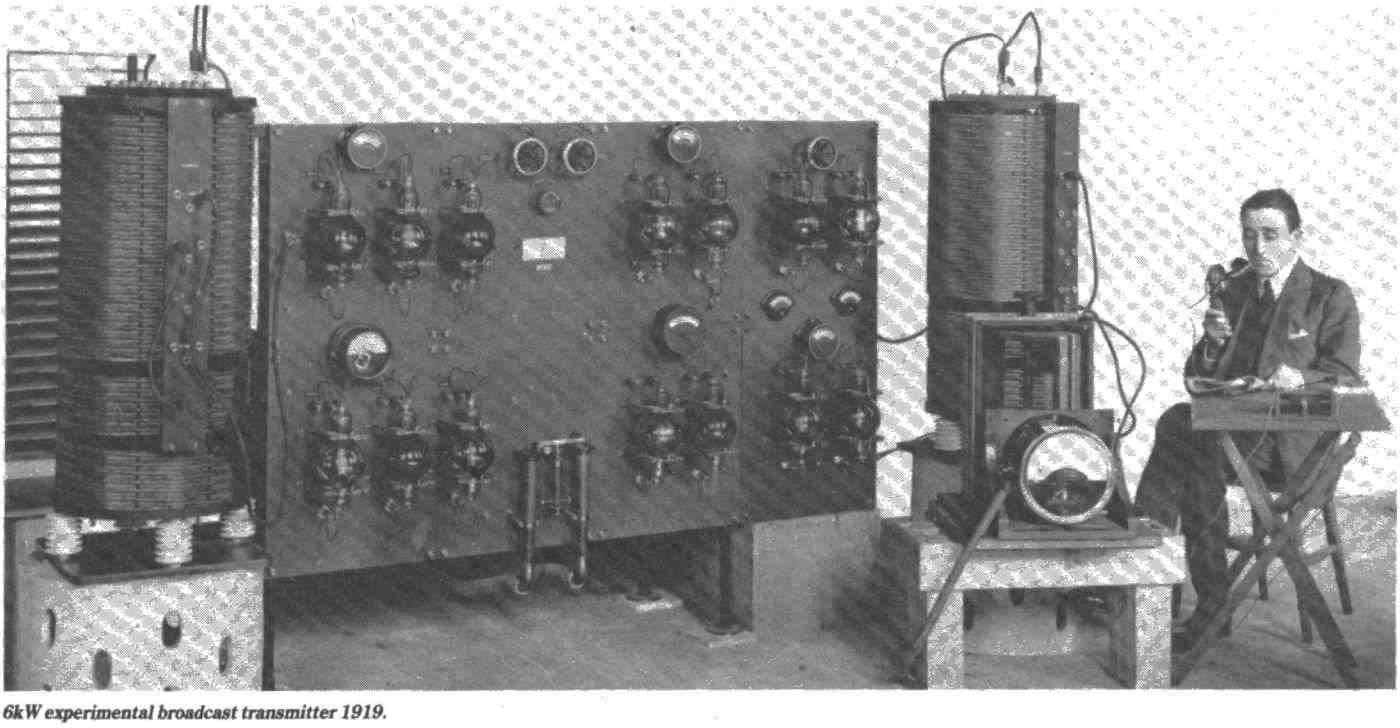


Photo supplied by The Marconi Co. Ltd.

6kW experimental broadcast transmitter 1919.

equipped with wireless telegraphy equipment. By 1926 the Marconi Company had fitted out 161 lifeboats in this way.

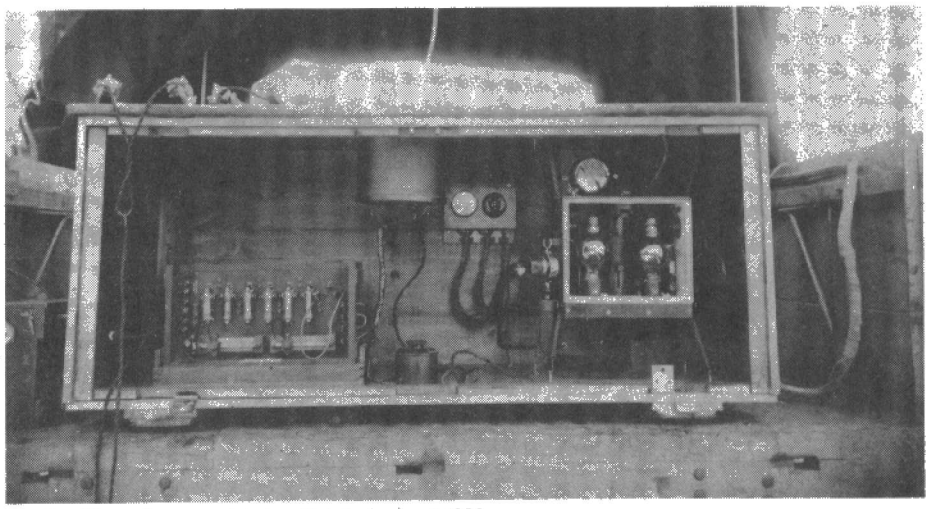
During the 1920's attempts were made to link the British Empire by wireless; in fact cable and wireless links existed side by side. In 1923 station GBR at Rugby was constructed, working on the incredibly low radio frequency of 16kHz. The aerial system required for working at such a long wavelength must be seen to be appreciated, consisting as it does of twelve 820 ft high masts! However, contact with any ship at sea or any part of the world at any time was guaranteed. In addition, a time signal and a news service was provided.

Radio telephony made great impact in that country of vast distances, Australia. Lack of an electric supply at isolated homesteads was overcome by means of pedal power. A generator driven by the adaptation of the motive parts of a bicycle produced the wherewithal that allowed contact between people well separated geographically. As well as the social and psychological implications of this communication link through the aether, a means of summoning assistance was readily available, leading to the Flying Doctor service that could be called up for help or just advice. Education for the children was another serivce that the radio provided, since attendance at a conventional school was often not possible. All this had happened by 1929.

Crimewaves

The first use of wireless for crime detection was the case already described of Dr. Crippen. However, this was a by-product of the public wireless service and not the result of a special police wireless network. Examples of the use of W.T. by the police can be found in Detroit (1921), the Metropolitan Police (1923) and the Lancashire Police, who used it during the General Strike of 1926. In 1928 the Brighton Police Force tested the first 'walkie-talkie' that could be carried by a constable, and it was in general use in the 1930's. But the number of available channels was limited by the bandwidth restriction imposed by broadcasting on the medium waveband.

Experiments in 'broadcasting', that is transmitting material for information or pleasure to a wide audience, had taken place before the First World War, but it could only have any real value when large numbers of people could receive the transmissions. The real beginnings of radio for entertainment in the home can probably be traced to the end of the 1914-18 war, when the factories, so long occupied with making instruments for war, were glad of the chance to do something for peace. Also the triode valve was showing its potential, although it was still very expensive. In the USA the Radio Corporation of America (R.C.A.) was formed out of the radio interests of a number of large companies, including Marconi's Wireless Telegraph Company of America. Soon the Americans were 'on the air' with station KDKA from Pittsburgh, transmitting to an estimated 5,000 - 10,000 listeners. By May 1922, there were three-quarters of a million receivers in use in the USA. It may not be easy to re-capture now the excitement of listening to 'the wireless', so used have we become to more and more breathtaking technological advances that no longer seem breathtaking!



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The first Marconi commercial air radio telephone set 1920.

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WIRELESS RECEIVERS
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PANELS ONLY.
Two Valve ..£12 0 0
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This very handsome Pedestal Cabinet fitted with New Model A. J. S. Four-Valve Receiver, H. and L. Tension Batteries, and A. J. S. Loud Speaker, the horn of which matches the wood, as supplied complete with all accessories ready for use in
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In the boom time of the 1920's even famous names of the motorcycle world diversified into the wireless world.

British Broadcasting

The birth of broadcasting in Britain was a much more restrained affair than in America, largely due to governmental opposition. However, start it did and, in 1919, Marconi's set up an experimental R.T. station at Ballybunnion in County Kerry, to communicate with Nova Scotia. Well, this may not sound much like broadcasting but it did lead to a 6kW transmitter being built at the company's Chelmsford works. The frequency of operation was 120kHz and test transmissions resulted in an increase in power to 15kW. The station, now fully licensed, went on the air on the 23rd February 1920, and introduced the first regular daily broadcast service in the world. It broadcast two half-hour programmes of speech, music and news every day. Although many who broadcast in the early days were just station staff, a notable exception to this

Photo supplied by The Marconi Co. Ltd.

Listening to the wireless 1922.

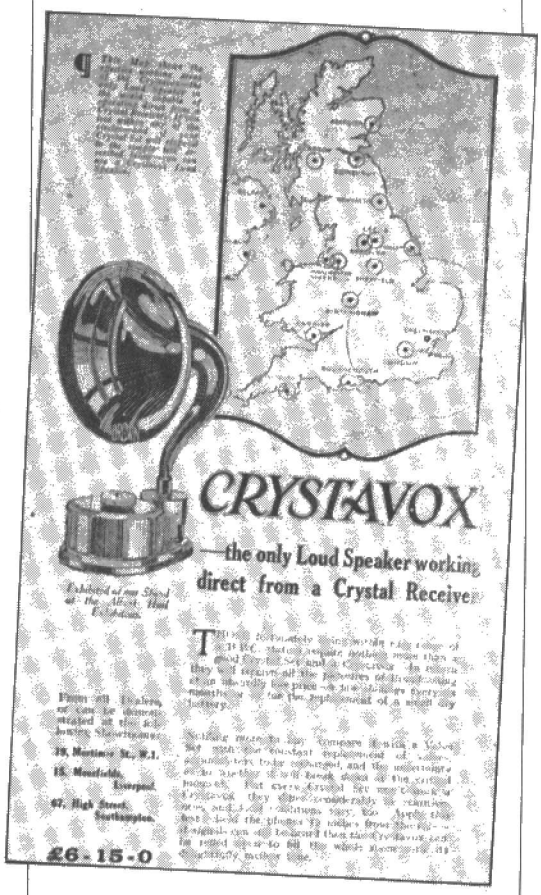
was the occasion when Dame Nellie Melba broadcast on the 20th June 1920, at the direct instigation of Lord Northcliffe, then owner of the Daily Mail and one of the most enterprising newspapersmen ever. However, in November 1920, short-sighted bureaucracy triumphed and the British station went off the air, the official view being, that the use of wireless purely for entertainment was trivialising a national asset!

Enthusiastic Amateurs

Other aspects pertinent to the total picture of what was happening in wireless at the time included the beginnings of amateur wireless and of a national industry to serve a public becoming more and more excited by the new diversion.

After World War One, there were many ex-servicemen who had picked up some knowledge of wireless and were reluctant to relinquish it on returning to civilian life. They therefore became enthusiastic amateurs and, to cater for them, courses were run at Marconi House and Crystal Palace. The first wireless clubs were formed and, as a result, much experimenting went on.

Following the first Chelmsford transmissions, an industry began to emerge whose aim was to produce receivers for an anticipated market of eager listeners. As interest grew so did the number of manufacturers. By the end of 1922, when the BBC was in existence, there was a flourishing radio industry in this country. But what of the government ban?



Advertisement in 1923 for the Crystavox.

In December 1921, a petition signed by over 3,000 members belonging to 63 societies was presented to the Post Office to demand the resumption of the Marconi Company's R.T. broadcasts. The Postmaster General, early in 1922, told the Marconi Company that a fifteen minute programme of music could be included in their weekly half-hour calibration transmission. A hut at Writtle, near Chelmsford, still in existence but now used as a sports pavillion, housed station 2MT. The first transmission went out on the 14th February 1922. A twenty-five minute broadcast was made every Tuesday evening. Soon after Writtle went on the air, the Marconi Company were issued with a licence for a broadcast station to be set up in Marconi House. This was the famous 2LO, which first broadcast on the 11th May 1922. At first only short speech transmissions were made but, from the middle of 1922, concerts and musical evenings were broadcast. Because of the low power of the first transmitter (only 100W) range was limited to about 40 miles for valve receivers, and 15 miles for crystal sets. But, when the power was increased to 1.5kW, the station could be heard in most parts of the country. Other companies involved in setting up stations were Western Electric and Metropolitan Vickers. The former set up a station, 2WP, in Oswaldstre House, Norfolk Street, London; the latter set up their station, 2ZY, in Manchester, where

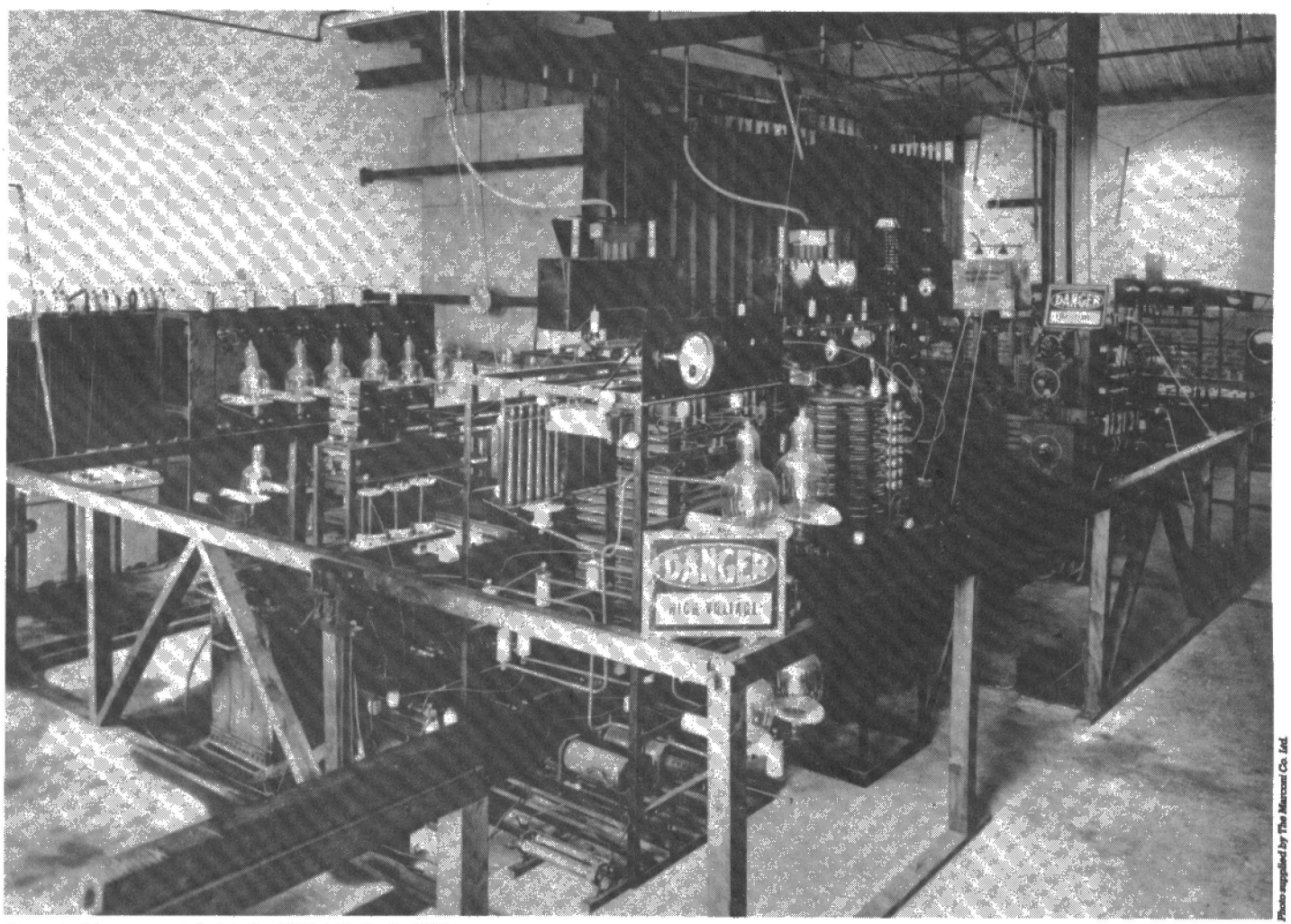


Photo supplied by The Marconi Co. Ltd.

Marconi wirless transmitting station 5XX at Chelmsford 1924.

they also manufactured their 'Cosmos' crystal set, costing £4 10s (£4.50) and a two-valve receiver costing £26 10s (£26.50). However, all broadcasting in the UK came under the corporate cloak of the BBC, when it was formed in October 1922, the participating companies being British Thomson-Houston, General Electric, Marconi, Metropolitan Vickers Electric, Radio Communication and Western Electric.

The Crystal Set

An obvious advantage of the crystal set was its cheapness and simplicity and its freedom from batteries. It was, however, vital to get the best signal possible and this meant rigging up a long aerial, as high as possible, and also providing a good earth, either by a connection to the domestic plumbing or to a metal plate in the ground, which then had to be kept well watered to maintain a good contact, one of the gardener's extra duties! A number of different circuits were in use, the tapped inductance circuit being shown in Figure 2.

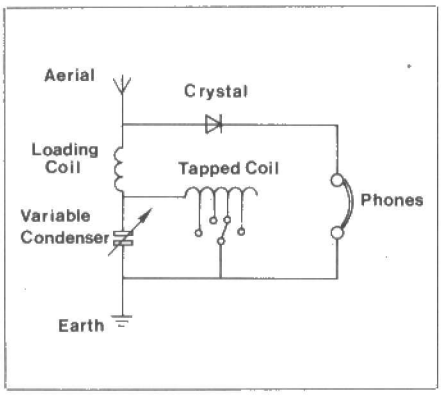
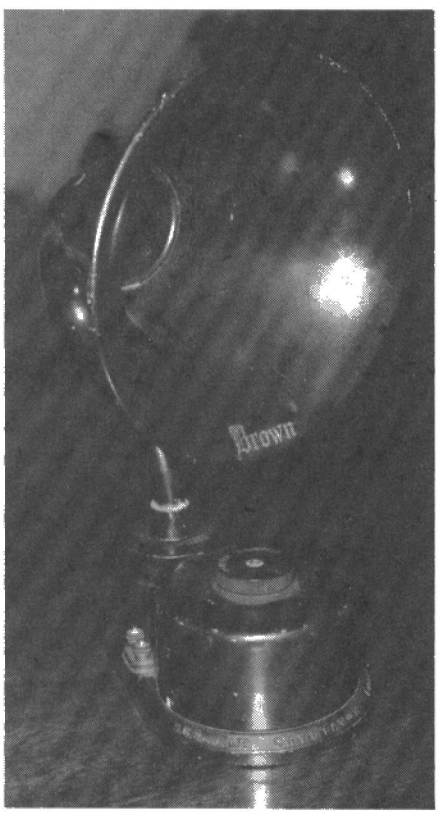


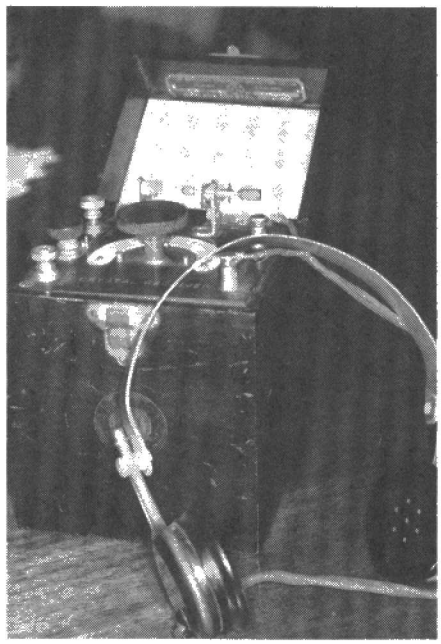
Figure 2. Circuit of a typical 'tapped inductance' type of crystal set of the 1920's.



The Crystavox.

B.T.H. RADIO VALVES
B3.
B4.
B5.
B6.
R.
B3 Valve ..... Price 21/-
Filament Volts ..... 9 volts.
Filament Current ..0.06 amps.
Anode Volts ..... 20-80 volts.
B4 Valve ..... Price 30/-
Filament Volts ..... 6 volts.
Filament Current ..0.25 amps.
Anode Volts ..... 40-100 volts.
B5 Valve ..... Price 25/-
Filament Volts ..... 3 volts.
Filament Current ..0.06 amps.
Anode Volts ..... 20-80 volts.
B6 Valve ..... Price 35/-
Filament Volts ..... 3 volts.
Filament Current ..0.12 amps.
Anode Volts ..... 60-120 volts.
R Valve ..... Price 12/6
Filament Volts ..... 4 volts.
Filament Current ..0.63 amps.
Anode Volts ..... 30-60 volts.
These five B.T.H. Valves meet every possible requirement of the ordinary listener-in and the serious experimenter. The R Valves can be used in any position, for detection or high or low frequency amplification, while the B4 and B6 Valves are intended primarily for low frequency power amplification.
Be sure your next valve is a B.T.H. Valve. Look for the initials "B.T.H." which are the sign of high quality, and for the silvered bulb which denotes a perfect vacuum.
Obtainable from all Electricians and Wireless Dealers.
The British Thomson-Houston Co., Ltd.
Works: Coventry.
Offices: Crown House, Aldwych, W.C. 2
B T-H

Some 1920's radio valves from a famous maker, including the renowned R type.



BTH crystal receiver with earphones.

The basis of all circuits was the rectifying action of the point contact formed between a wire 'cat's whisker' and galena crystal (in fact, a semiconductor diode even if it wasn't known as such). A resonant circuit tuned by either variation of capacitance or inductance selected the required station. The audio output from the 'diode' was enough to drive a pair of high impedance phones in a high signal strength area. This meant that 'listening-in' in the family circle tended to be done in turns. To overcome this estrangement from the rest of the domestic scene, one solution was offered by the company of S.G. Brown, with their 'Crystavox'. This was an amplifying loudspeaker with a horn output. It was entirely electromechanical and took its power from a 6V battery, its current consumption being a mere 15 - 20mA. It did allow some degree of communal entertainment, though it literally demanded a close family circle as well as a very good signal from the crystal set feeding it. Apart from complete receivers, manufactured in some cases by some unlikely makers, for example, A.J.S., the well known motorcycle firm, components were available as well as kits of parts for people to make their own crystal sets.

Valve Receivers

The valve receiver naturally had real gain and a potentially greater power output than a crystal set, provided it included two or more valves. A 'one-valver' drove 'phones in the manner of the crystal set but had greater range. Even so very good aerials were needed, usually fixed to the wall of the house or the chimney, on a mast system, rather like the wireless aerials in ships. The power for the receiver commonly came from a 90V high tension battery and a 2V lead-acid accumulator, which needed periodic re-charging at the local garage or wireless shop. Such items were expensive, as were the valves, which also had a very limited life. The design of many sets were curious by modern standards, the valves often being mounted on the top in full view. At least this allowed them to be changed easily, and also let the occupants of the room read by the light of their 'bright emitter' filaments! But of course the real advantages of valve receivers were the ability to listen to more distant stations as well as being able to drive a horn loudspeaker so that everyone in the family could listen at once.

Styles in receivers changed in the 1920's, so that the design co-ordinated better with current furniture styles. The valves retreated to their rightful place inside, on a proper chassis, and the horn loudspeaker began to be replaced by paper cone types. These appeared in elaborately fretted cabinets. As enthusiasm for the wireless grew and its place in the home became accepted - as a necessity and not just a luxury - so it matured in design both functionally and aesthetically. Although a lifeline to many people in World War Two, perhaps the 1930's was the 'heyday' of the wireless as a means of entertainment, but more of that anon.

NOTCH FILTER

by C.S. Barlow
& Ms M.A. McCarthy

Introduction

Frequently, when listening to radio stations on the short wave bands, an annoying interfering signal causing a loud whistle is also received. At the same time the source of these interfering signals is varied. Some are the by-products of local electronic equipment, i.e. digital and radio frequency (computers, televisions, etc.). Others are more deliberate and more distant; jamming, military and commercial. Most communications receivers have a good quality intermediate frequency (IF) filter and some have a bank of filters to select the appropriate bandpass to suit the transmission mode being received. However, if the interfering signal is within the passband of the IF filter the signal will be heard. To remove this unwanted signal requires the small part of the IF containing it to be notched out. If the notch is not very sharp then too much information will be removed and the wanted signal will suffer. In addition, the depth of the notch is important because the more that can be removed, the less it will be heard once it is demodulated into an audio signal.

The inclusion of a Notch Function is quite usual on more expensive communications receivers, but if yours has not got this facility, then adding it could prove to be difficult and costly - especially if a mistake is made during the modification!

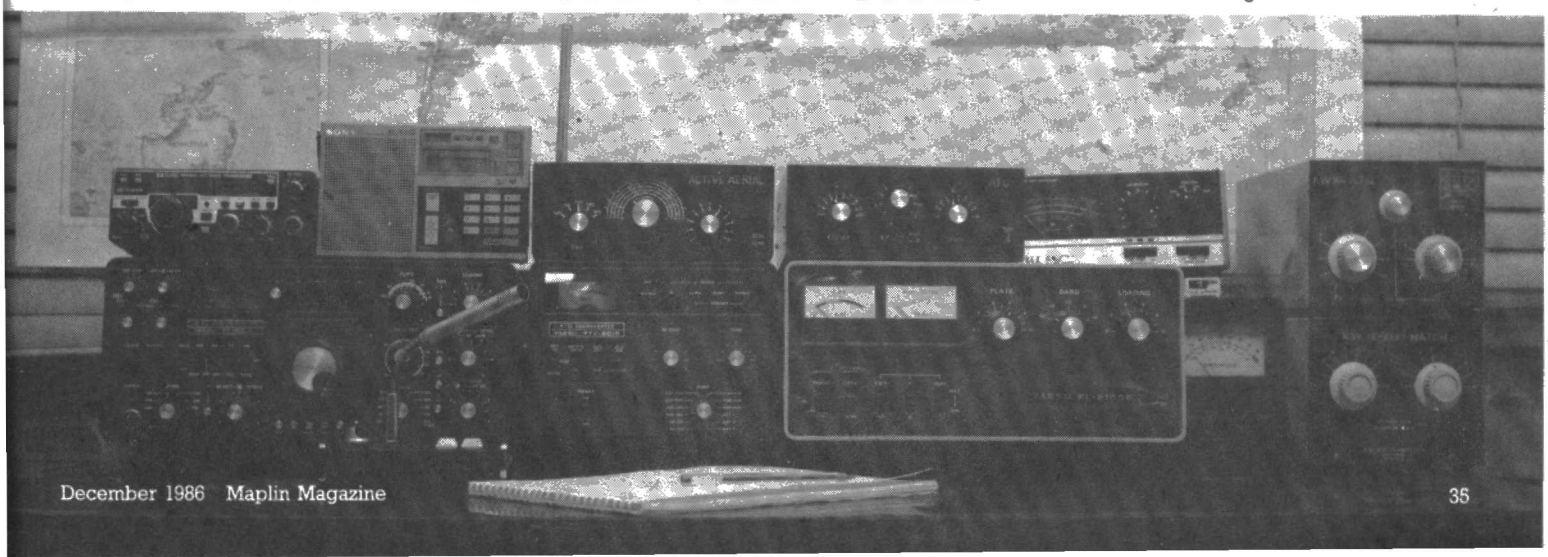
The reason for the IF normally being notched, is because it is easier to obtain a narrow response at this higher frequency than at the lower audio frequency of the demodulated signal. However, it is possible using modern circuitry to produce a narrow and deep notch at audio frequencies. This means no modifications are necessary to your receiver - since the Notch Filter can be placed between the receivers loudspeaker or headphone output and the loudspeaker or headphone itself, see Figure 1.

The filter described in this article has two functions; the first is to notch out an interfering tone within a tuneable range of 200Hz to 4.5kHz, and secondly, over the same range, it can peak or boost the wanted tone being received. The reason for the second function is when listening to a morse or data transmission, you can enhance the signal by boosting just the wanted tone. Even if your receiver already has an IF notch filter, an extra audio one is a useful addition and the two filters will work well together.

SPECIFICATION OF PROTOTYPE

All readings were taken under the following conditions:

Power supply voltage:	+12V	THD:	0.04% at tape output
Frequency input:	1kHz	Power output:	1 Watt peak
Loudspeaker:	8Ω	Quiescent current:	47mA
Input level:	0dB	Current at full output (8Ω):	200mA
Results noted:		Current at full output (4Ω):	400mA
Tuning range of filter:	200Hz - 4.5kHz	DC input voltage range:	
Notch mode:	−55dB	Minimum:	+9V
Peak mode:	+6dB and −12dB off peak (difference of 18dB)	Maximum:	+14V



Circuit Description

In Figure 2, the DC power input for the circuit is applied to SK1 and passes through the 500mA fuse, FS1. S1a switches the power and if the polarity of the voltage is accidentally reversed then D3 will be forward-biased and blow the fuse. The reason being that it is cheaper and easier to replace a blown fuse than an expensive collection of IC's! When the power is applied in the correct polarity (+V to FS1 and −V to the ground), D3 will be reversed biased and have no effect. Then LED1, the power indicator, will light. The main power supply de-coupling is provided by C15, a 470μF capacitor. Because this notch filter will normally be placed between the speaker and the output of your communications receiver, the power switch incorporates a bypass function, switching the speaker directly to the communications receiver when the power to the unit is turned off. This function is provided by S1b and S1c.

For the circuit to function correctly a half supply reference is necessary. This is provided by half of IC1. The voltage reference applied to the input of this Op-Amp is derived from the two resistors R12 and R13 which form a potential divider. The Op-Amp is merely used as a zero gain buffer to provide a low impedance half supply, its output being de-coupled by C4 and C5.

The other half of this IC is used for the overload indicator. The audio signal going to the output IC, IC2, is coupled by C8, R10 into a rectifier circuit consisting of diodes D1, D2. The combination of R1, C1 and R2 smooth out the signal into a DC level. When this voltage exceeds the reference voltage set by R3 and R4, pin 1 of IC1 goes high. This voltage is fed via R7, R8 to the base of TR1 which turns on the transistor causing a current to flow through LED4 and so making it light up.

The signal from your receiver is applied to SK2 and the loudspeaker is then connected to the output socket SK5. These sockets are the quick-release terminal type for ease of connection into your speaker line. On each socket one terminal is the ground and the other the signal, i.e. black for ground and red for signal. If the loudspeaker is connected to the input socket by mistake and the receiver to SK5, then while the unit is switched out the sound from the receiver will still be heard, but when it is switched in, no sound will come through.

When switched in the signal is applied to the input circuits via C6, a 10μF capacitor. The input impedance at this point is quite high so it can be driven from the headphone output. This may be necessary if your receiver has an internal speaker and no provision for an external one. On some receivers it is necessary to present the correct low impedance at the speaker output, so an optional load resistor of 10Ω can be switched in. This is usually only necessary on older valved receivers with transformer coupled output stages. When using the load switch S2 on modern transistor receivers, the effect of the switch will be negligible. On valved receivers the switch will have the effect of reducing the volume while allowing the volume control to operate over its correct range and preventing any possible damage to the output stage of the receiver.

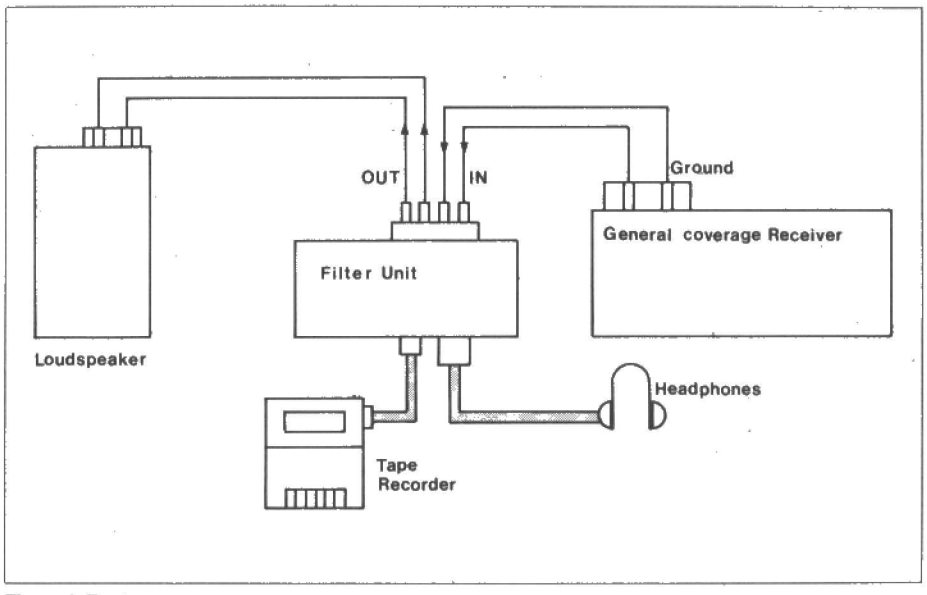
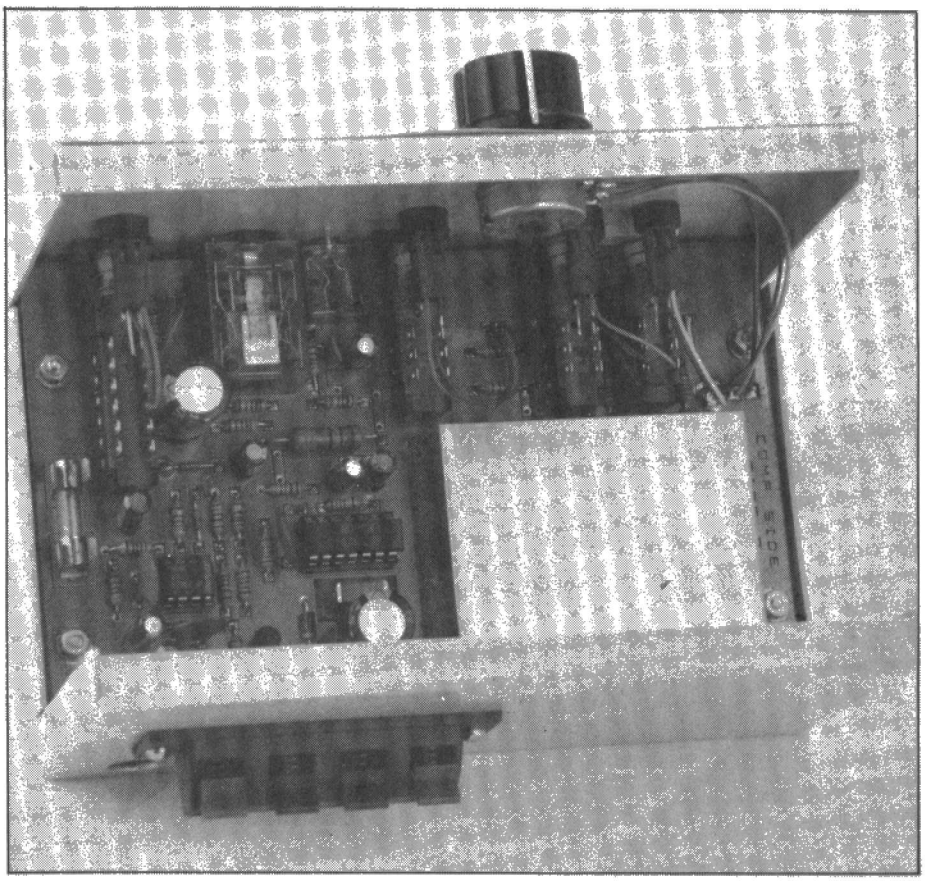


Figure 1. Equipment set-up.

Because the input can be at a relatively high impedance, a small value ceramic capacitor, C7, is used to de-couple any stray RF signals that may be present if using the unit with transmitting equipment. The audio signal then passes to the filter IN/OUT switch S4. In the OUT position the signal is connected directly to the sound output stage IC2. When pushed IN it selects either the Notch or Peak mode of operation set by S3, OUT for Notch or IN for Peak. This switch also controls which LED is lit; LED2 for Notch or LED3 for Peak. Because of the high input impedance of the filter it can be left permanently connected to the signal input line without adversely affecting the direct signal.

The signal enters the filter through C17 and an attenuator which comprises R48, R25, R26 and is connected to the first half of IC4, an MF10CN. The Notch output of IC4 is fed to one half of IC3, and its output is then passed to the second half of IC4 via another attenuator circuit comprising R32, R36, R37. The final Notch output is at pin 7 of IC3. The effect of using the two halves of the MF10CN in this way is to increase the depth of the Notch, thus reducing the interfering signal to a minimum.

The first part of the MF10CN also produces the Peak filter output and this signal is then fed via R41 to the input, pin 2 of IC5. A small amount of Notch signal is also fed to pin 2 via R40, (the effect of which is to allow a low



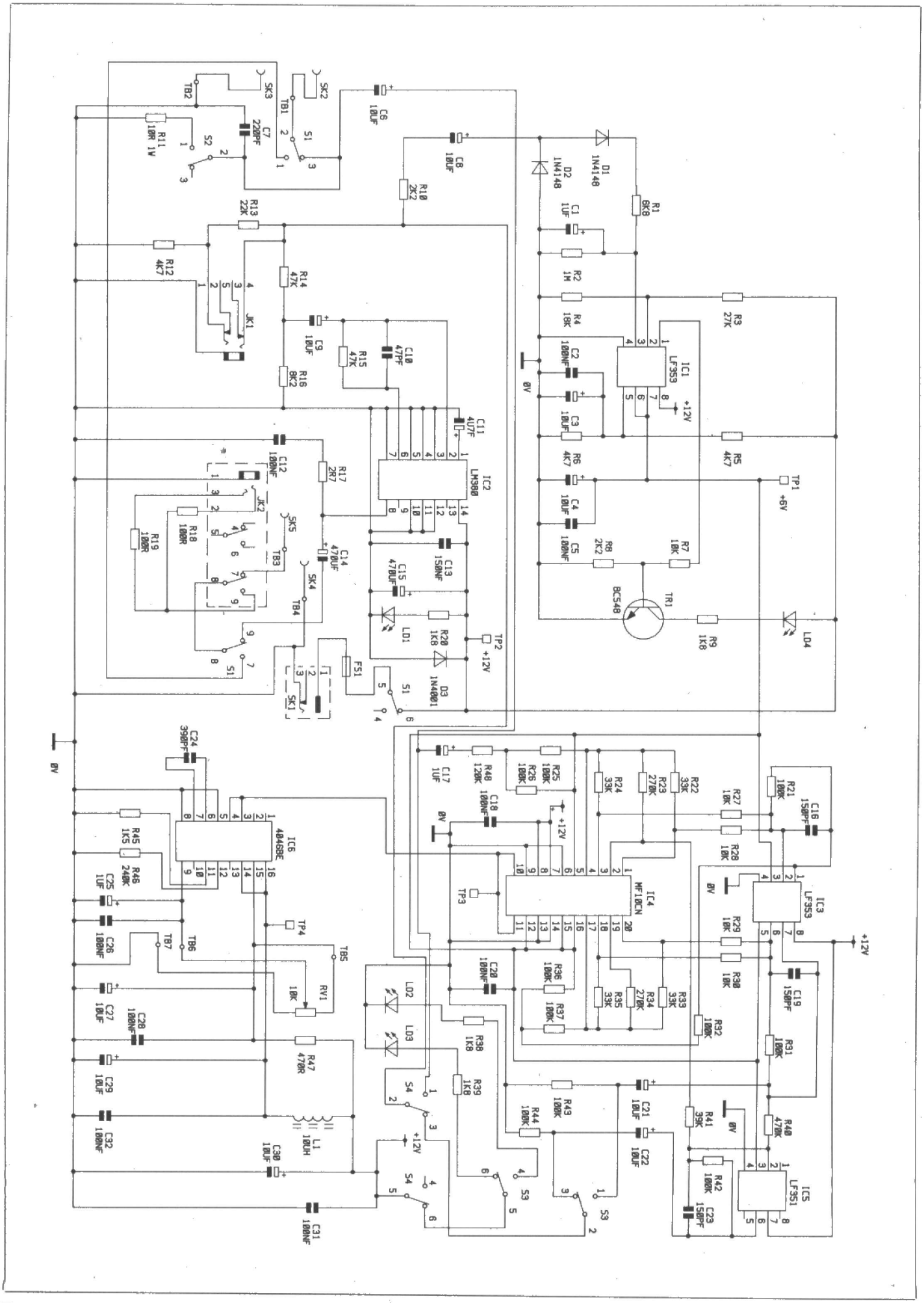


Figure 2. Notch Filter circuit.

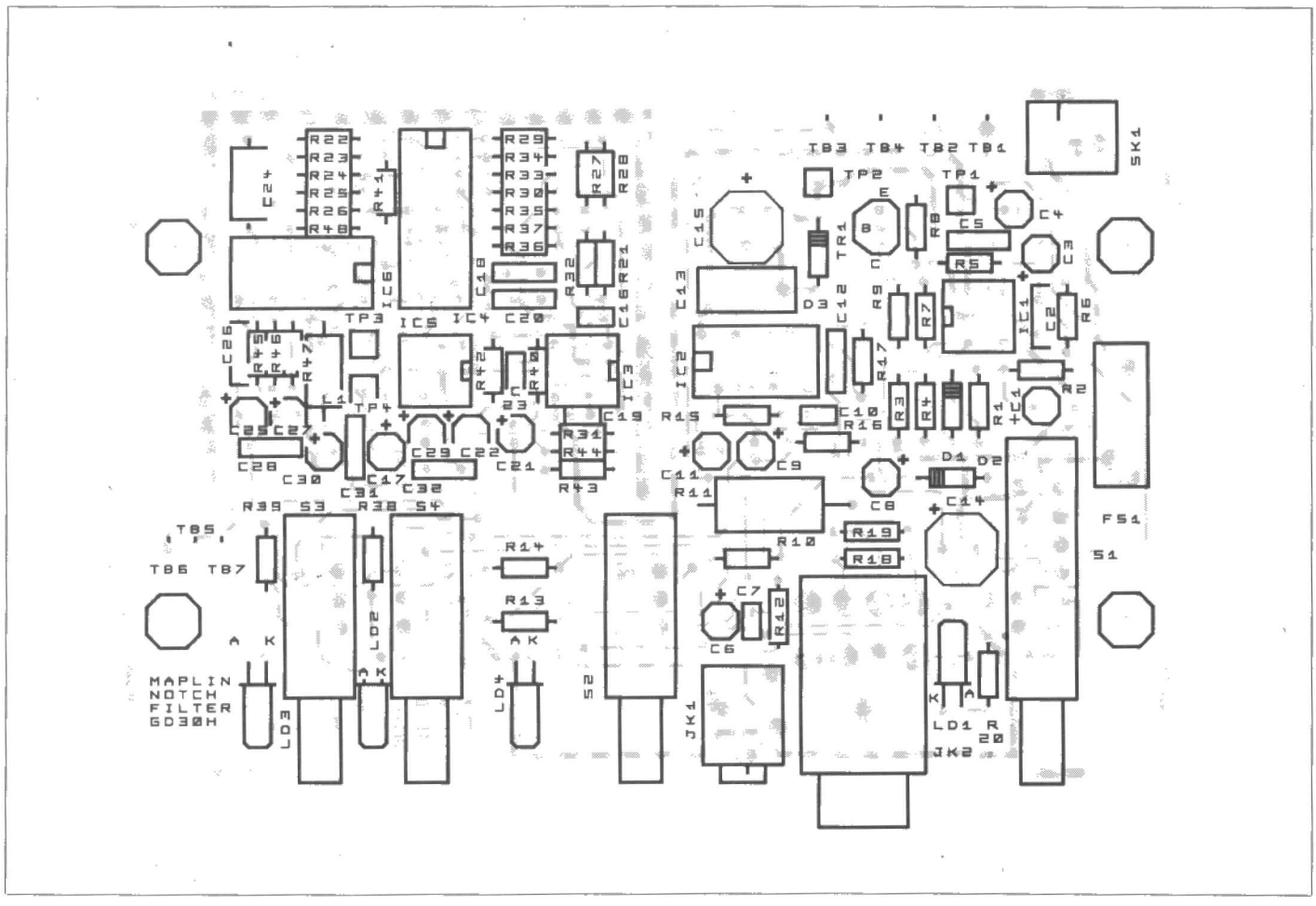


Figure 3. How the pcb is laid out.

level of the whole audio spectrum to be heard, however, the signal at the frequency of the Peak filter will be much higher), where it is present on pin 6 of IC5. For the MF10CN to Notch or Peak at the desired point, a variable frequency square wave must be applied to the clock input, pins 10, 11.

The clock signal is generated by IC6, a 4046BE, running at a frequency range of between 20kHz and 400kHz. These frequencies are one hundred times the audio frequency that the MF10CN is required to handle in the Notch or Peak modes. The tuning of the clock is set by RV1, the 10k linear potentiometer, which alters the voltage at pin 9 of IC6. When this voltage is at its minimum setting the frequency output of IC6 will be at its lowest. The maximum clock frequency will occur at the highest voltage setting and this voltage reference is generated by a zener diode inside the 4046BE and appears on pin 15. The power supply for IC6 is fed via an RF choke, L1 and de-coupled by C29 and C32, preventing any of the high frequency clock signals straying on to the power supply rail. Screening is also necessary around this circuit to prevent the radiation of harmonics being picked up by local radio receiving equipment.

The signal going to the output stage is split off to JK1, the stereo 3.5mm jack socket, which provides the tape output signal. Although a stereo socket is used the output will be mono but at two different levels. Depending on your tape recorder, one or the other should be suitable. The sound output stage IC2, the LM380, is capable of driving 4 or 8Ω impedance loudspeakers or headphones. The headphones can be wired for mono or stereo and when plugged into the quarter inch jack socket, JK2, the speaker is automatically switched out.

PCB Assembly

The PCB is a double-sided, plated-through hole type, chosen for maximum reliability and stability. However, removal of a misplaced component is quite difficult with this kind of board so *please* double-check each component type, value and its polarity where appropriate, *before* soldering! The pcb has a printed legend to assist you in correctly positioning each item, see Figure 3.

Do not fit the metal screening or the IC's until the initial testing stage!! The sequence in which the remaining components are fitted is not critical. However, the following instructions will be of use in making these tasks as straightforward as possible. It is usually easier to start with the smaller components. Begin with the resistors as usual, but ensure that R11, the 10Ω 1 Watt speaker load, is mounted at least a quarter of an inch above the circuit board to allow good ventilation. Next mount the ceramic, polyester, polystyrene and electrolytic capacitors. The polarity for the electrolytic capacitors is shown by a plus sign (+) matching that on the pcb legend. However on some capacitors the polarity is designated by a negative symbol (−) in which case the lead nearest this symbol goes *away* from the positive sign on the legend. The diodes D1-D3 have a band at one end. Be sure to position them according to the legend, where the appropriate markings are shown. When fitting the RF choke L1, be very careful while bending the leads in order not to damage this component. There is only one transistor in the entire project, but you must carefully match the case to the outline shown on the legend. Next install the switches making certain that they are pushed down firmly on to the surface of the pcb. *Do not fit* the push buttons onto the switches at this stage.

Discard the fixing nuts on JK1 and JK2, installing the connectors as carefully as you did the switches and firmly against the board. Next comes the DC power input socket SK1 which is similarly mounted. The fuse clips are then installed at the two square markings, FS1. The final PCB mounted sockets are for the IC's. Ensure that you fit the appropriate IC holder in each position, matching the notch with the block on the legend. The remaining components are connected to the circuit board by wires. If you purchase a complete kit from Maplin it should contain a length of ribbon cable. No specific colour has been designated for each wire connection, it is entirely up to you. The use of coloured ribbon cable is to simplify matters, thus making it easier to trace separate connections to off-board components, just in case there is a fault in any given part of the circuit. Strip off from the main group whichever colour you prefer for each installation. Using three pieces of wire 100mm in length, connect RV1 to the PCB as shown in Figure 4. The LEDs are connected in circuit by the 2-pin 'Minicon'

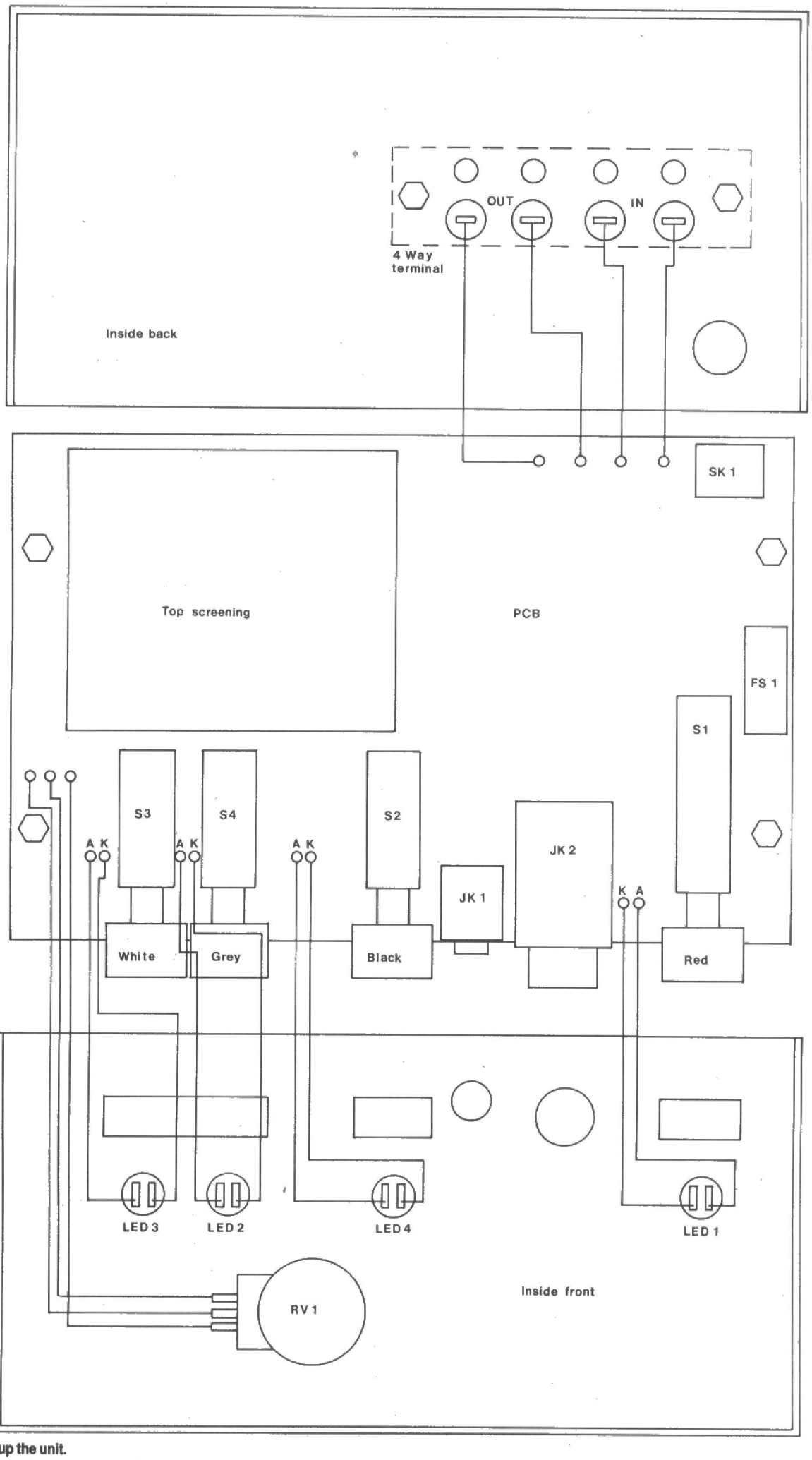


Figure 4. Wiring-up the unit.

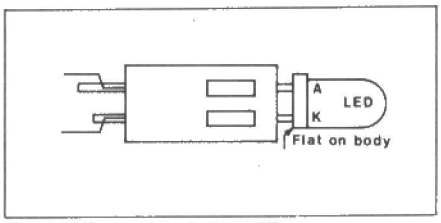
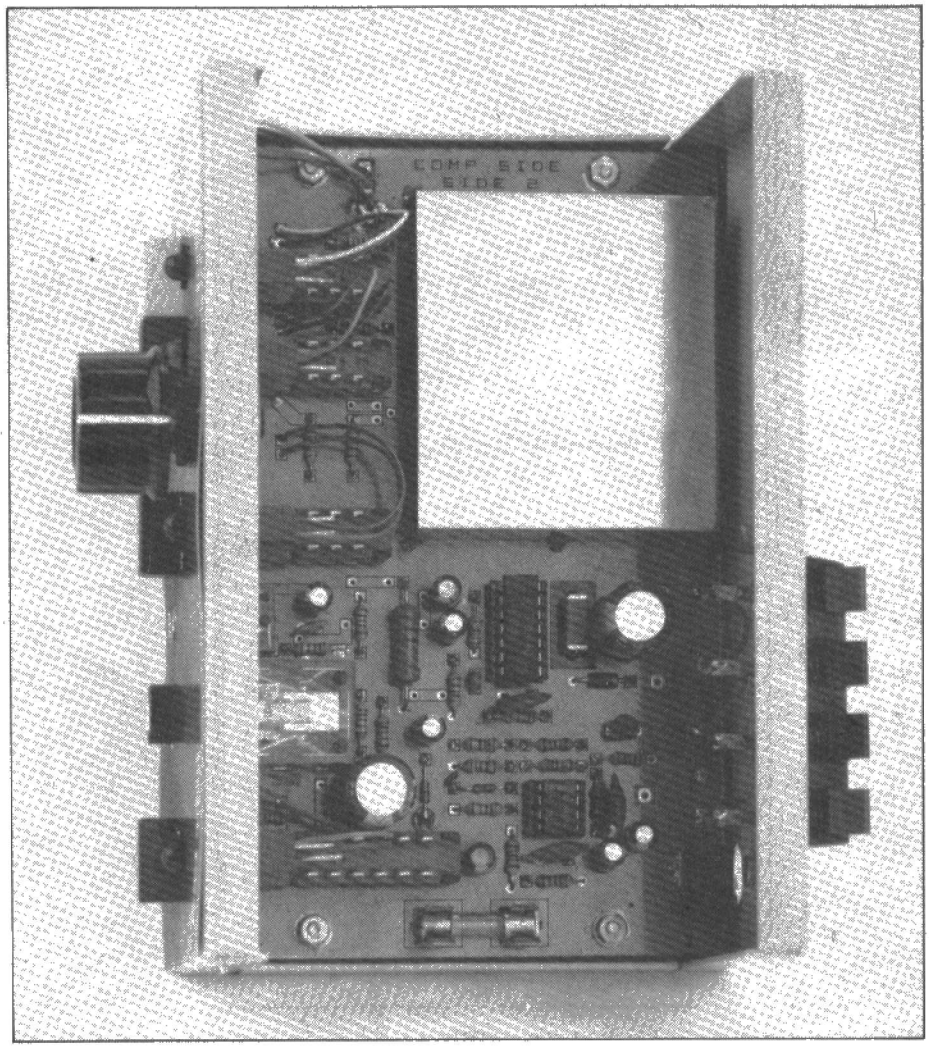


Figure 5. Connecting to the LED's.

plugs which are wired to the pcb as shown in Figure 5. The method of installing and soldering the wire to the 'Minicon' terminal is shown also in Figure 6. These wires are of equal length and should be no longer than 65mm. The LEDs have a short lead and a flat edge on one side of their case to identify the cathode (k). When inserting them into the 'Minicon' plugs make sure that they are the correct way round, otherwise they will not light. However, under these conditions no damage will occur and it is a simple matter to remove and reverse the 'Minicon' plug. Finally, add four 40mm pieces of wire for connection later to the terminals mounted on the back of the metal case.

This completes the assembly of the pcb and you should now check your work very carefully making sure that all the solder joints are sound. It is also very important that the track side of the circuit board does not have any trimmed component leads standing proud by more than 2mm, especially around the area where the bottom screening plate is to be fitted later, as this may result in a short circuit. To prevent this, cut all component leads flush with the solder. Further information on soldering and assembly techniques can be found in the 'Constructors Guide' included in this kit.



Initial Testing

All the initial tests can be made with a minimum of equipment. You will need a Multimeter and if possible, a Frequency Counter or Oscilloscope. The readings were taken on the prototype using a Digital Multimeter and some of the readings obtained may vary slightly depending on the type of meter you use.

Carefully lay out the pcb assembly on a non-conductive surface, such as a piece of dry paper or plastic. Position the tuning control RV1 and the LED's so they are clear of the circuit board and make sure the four wires at the rear of the board are also clear. The DC input socket SK1 is a type commonly used on Japanese radio equipment, where the centre pin is the positive connection and the negative contact is the spring-loaded tag. The first test is to measure the resistance at SK1. Ensure that the 500mA fuse is fitted. When the unit is switched off, no resistance should be detectable either way round on the positive or negative pins of SK1. But when the unit is turned on, by S1, then in one direction you should get a reading in excess of 3kΩ and when the test leads are reversed, a lower reading in the order of 500Ω should be present. These readings are due to D3, the component which protects the rest of the circuit from damage.

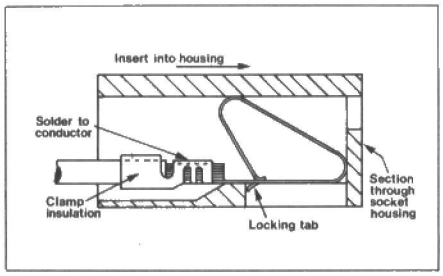


Figure 6. Minicon terminations.

In all the following tests it will be assumed that a power supply of +12V is being used, although it can be as low as +9V for a reasonable performance. The power supply should never exceed +14V, therefore it is advisable to use a 12V battery for portable use, or a stabilised power supply for home use. Before connecting the power supply to the unit, make sure all the switches are in their OUT positions and that RV1 is set fully anti-clockwise. Remove FS1 and connect the positive lead of your Multimeter to the fuse clip nearest to the power socket and the negative lead to the other clip. Select a suitable range on your meter that will accommodate a 500mA DC current reading. Now connect your power supply to SK1 and switch on. LED 1 should light and a current of 9mA should be observed. Switch off the unit and replace the fuse. Set your meter to read DC volts. All voltages are positive with respect to ground so connect your negative lead to a convenient ground point. When the unit is switched back on voltages should be present at the pins of the IC sockets according to the following table. *Remember* none of the IC's should have been installed at this time.

IC1	Pin 8 = +12V
IC1	Pin 5 = +6V
IC1	Pin 2 = +4.7V
IC2	Pin 14 = +12V
IC3	Pin 8 = +12V
IC4	Pin 7 = +12V
IC4	Pin 8 = +12V
IC5	Pin 7 = +12V
IC6	Pin 16 = +12V
IC6	Pin 14 = +12V
IC6	Pin 15 = +6.3V

The unit should now be switched off and IC1, the LF353, installed. Make sure that the marker for pin 1 aligns with the legend on the pcb. Turn on the power and measure the voltage at either pin 6 or 7. A reading of +6V should be present. Pin 1 should be at 0V and the overload LED should not light. Turn off the power once again and install IC's 3 and 5. Switch on and a reading of +6V should be found on pin 3 of both IC's. Pin 5 of IC3 should also show a reading of +6V. Next check pins 5, 12 and 16 for a reading of +6V. Switch off. If this test is successful then IC4, the MF10CN, and IC6, the 4046BE clock generator, should be fitted. Now switch on.

If you have an oscilloscope or a Frequency Counter, monitor the frequency of the clock at TP3. With the tuning control set fully anti-clockwise, a reading of

approximately 20kHz should be displayed and with the control set fully-clockwise the frequency should increase to approximately 450kHz. If you do not have access to this type of equipment, all you can test is the change in DC control voltage at pin 9 of IC6. With the control knob fully anti-clockwise, no DC should be present, but as you turn the control clockwise an increasing DC voltage should be observed. When you reach the end of the travel of the control a reading of approximately 6V should be seen. Switch off the unit and install the final IC which is the 'sound output' chip IC2, the LM380. Switch on and measure the voltage at pin 8, where a reading of approximately 6V should be present.

Now that all of the chips are fitted, repeat the current test by removing FS1, as described earlier. A reading of not more than 50mA should be drawn from the power supply. When switching on and off you may have noticed that LED4, the overload indicator, flashed on for a short time. Don't worry, because this is quite normal! When you push in S4, the filter IN/OUT switch, LED2 the Notch mode indicator, should light. When pushing in S3, the Mode switch, the filter is set to the Peak mode. LED2 should go out and LED3 light.

This completes the initial DC tests. In the next stage, two methods of final testing will be described. The first is for those fortunate enough to have their own test equipment, and the second, a simpler test, requires just the receiver and speaker system which is to be used with the unit.

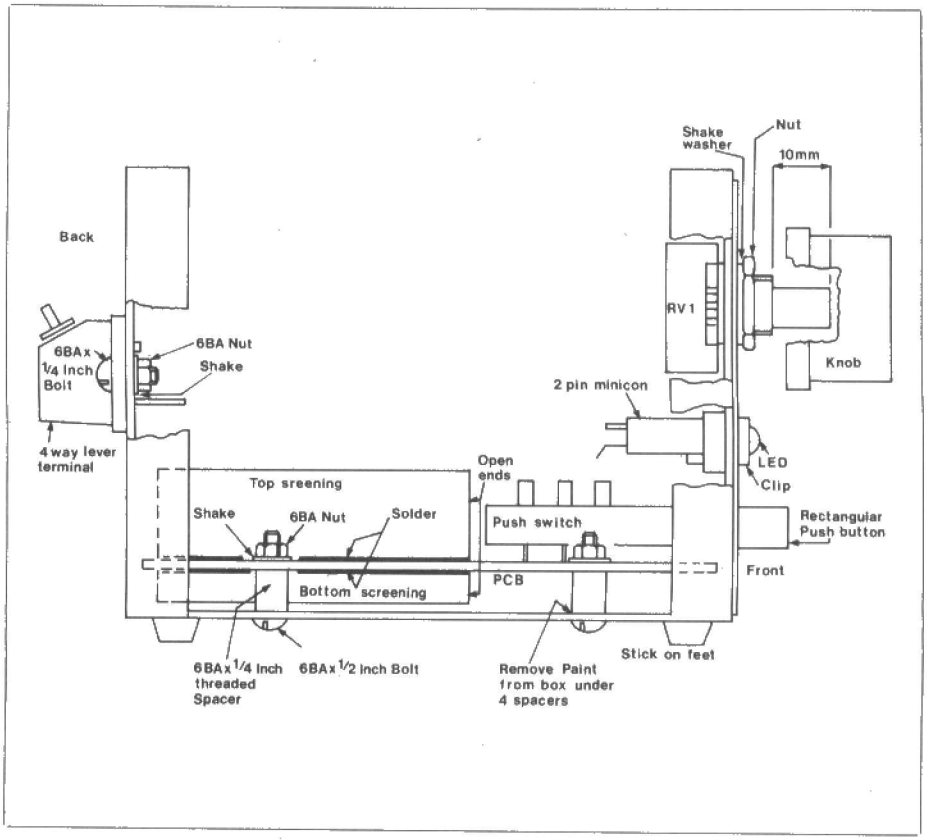


Figure 7. Final assembly details.

Final Testing

Before commencing either of the following testing procedures, set all the switches to OUT and RV1 to its full anti-clockwise position.

1) Testing with special equipment

Set your Audio Sine Wave Generator to give 1V r.m.s. at a frequency of 1kHz. Connect the output signal to SK2, and the earth to SK3. Now connect your oscilloscope probe to SK5 and its earth to SK4. Set your oscilloscope input sensitivity and sweep rate to show the direct signal passing through the unit. When you switch on, the signal amplitude and wave form should not alter significantly because the filter circuit is not engaged. The only active part is the output chip, IC2; but when the filter is switched IN, by pressing S4, again no change should be seen. The output amplitude will only drop when the tuning control is set to notch out the signal. If all is working well, the signal should virtually disappear from the screen and further increases in Y input sensitivity will be necessary to see the much weaker signal remaining, when fully notched out.

The opposite condition is generated when the filter mode switch S3 is pushed IN, giving a sharp peak at the tuning point of RV1. If the signal peaks too high then the overload LED4 will light and you will see clipping on the waveform. Under these conditions simply turn down the output of your signal generator until the LED goes out. These tests can be repeated over the frequency range of the filter.

2) Testing without special equipment

The exact point of connection of the filter to your receiver will depend upon the type you have. On most older, and some modern receivers, the speaker is separate and so the Notch Filter connects in-line with the speaker. Make sure the signal and earth wires are connected the right way round.

If your receiver does not have this facility, the headphone output can be substituted in most cases. If, however, you are using an older, valve type receiver then you should switch IN the speaker load with S2. This is not as important on modern transistor types and may only give a slight tonal change in sound quality, but the valve output stage must have a suitable load. Occasionally, when using the headphone output of the receiver, a large drop in volume may be detected when the load switch is engaged. This is the case for the Maplin 80m Receiver Kit (LK41U) which is designed for use with high impedance headphones (2k or greater). When the load switch is OUT the filter unit has an impedance greater than 2k and thus is well suited for use with this receiver. An added advantage is that now an 8Ω speaker or a pair of low impedance headphones can be used.

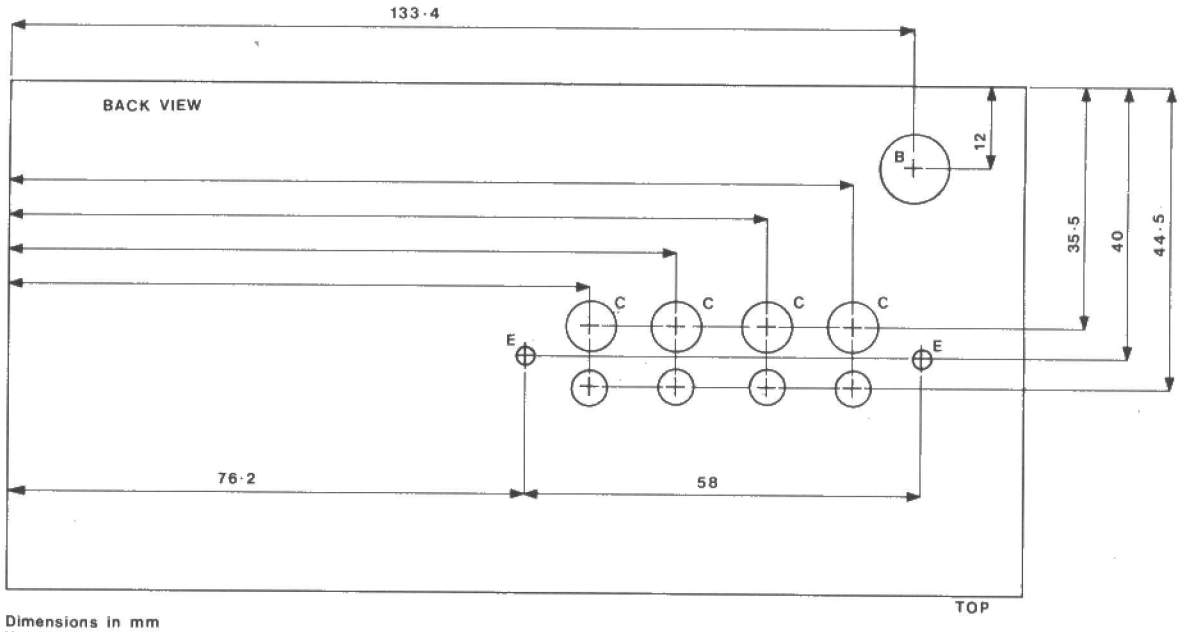
The testing is a simple matter of listening to a whistle or hetrodyne on the receiver which falls within the frequency range of the filter. In the Notch mode, when RV1 is swept over the offending tone, it should be greatly reduced in volume. In the Peak mode, it should be at a low level until you tune over the tone, then it will become louder. If, at any time during these tests, the overload LED should light, simply back off the volume control on your receiver until the indicator goes out.

This completes the testing of the unit and if all checks were successful you are ready to commence the final assembly.

Final Assembly

The only remaining items to be fitted onto the pcb assembly are the RF screening cans which solder to the earth planes as shown in Figure 7. Remove the LED's from the 'Minicon' plugs and set the pcb assembly aside temporarily. The metal case that the unit is designed to fit is the Blue Case type 212. It is supplied with an aluminium sub-chassis which should be discarded. The actual case is painted on all of its surfaces, so remove the paint from the inside at the positions where the 6BA threaded spacers are to be fitted, see Figure 7. This will ensure a good earth connection to the chassis. Follow the drilling instructions in Figure 8. When preparing the front of the case, the self-adhesive front panel can be used as a guide for checking the positioning of the holes. Having completed the drilling, at the same time clearing away any swarf, clean the front of the case and remove the protective backing from the self-adhesive front panel. Carefully position and firmly push down using a dry, clean cloth until it is securely in place.

Before installing the pcb assembly, fit the four threaded spacers. Carefully slip the board into place over the four fixing points and secure it with the 6BA hardware. Now fit the LED's into their clips, and push on their 'Minicon' plugs. Push the coloured buttons onto the switches as follows: Red on S1, Black on S2, Grey on S4 and White on S3.



Dimensions in mm
Hole Data
A 6·75
B 10·5
C 7·0
D 13·0
E 3·5
F 5·0

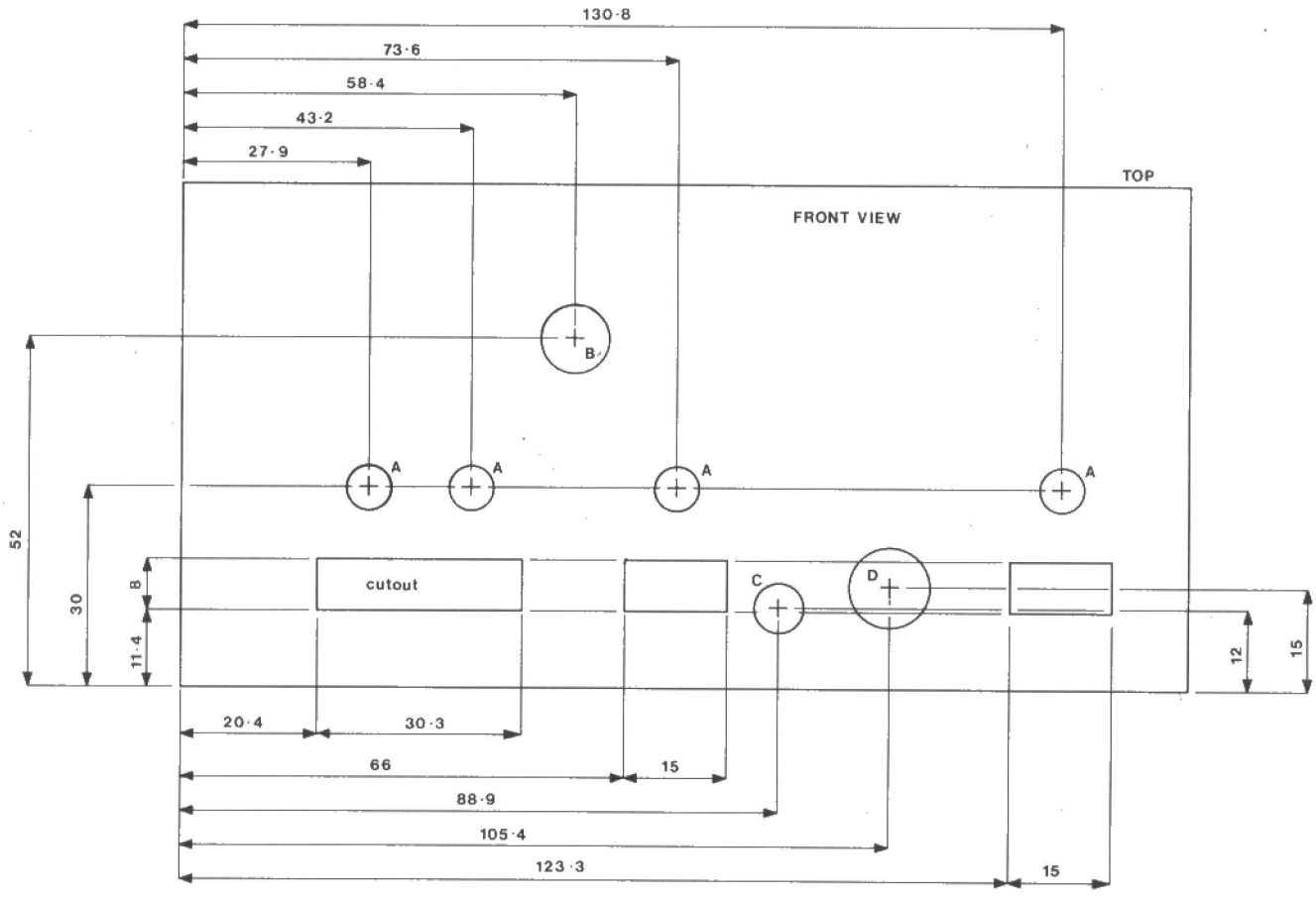


Figure 8. Case drilling details.

Next prepare the tuning control by cutting the shaft to a length of 10mm. Mount the control and secure the knob so that the pointer is at 0. Check that it travels smoothly round to the fully-clockwise position, without scraping on the front panel. This completes the front of the unit.

The only item to be fitted onto the back panel is the four-way terminal block for speaker and receiver connections. Wire up the terminals as shown in Figure 4. Check that the hole for the DC input socket SK1 is in line, in order that the DC plug will pass through it. Before putting on the lid of the case, connect up and test the unit just to make sure everything is still functioning. If all is well, fix the lid using the screws provided with the case and finally stick on the four rubber feet as shown in Figure 7. The unit is now ready for use.

Using the Unit

When installed and working, the best way to familiarise yourself with the functions of the unit is to tune around on your receiver for a speech (SSB or AM) or morse (CW) signal. When you find a speech signal with some interference on it, set the filter to its Notch mode and slowly adjust the filter tuning control. If the interfering signal is simply a single tone then you should be able to notch it out well below the speech level. If, however, the interference is of a more complex structure a compromise has to be reached and just the loudest component removed.

On CW transmissions, the Notch Filter is used in a similar manner, however the Peak mode may be more appropriate. If listening to a speech transmission whilst in the Peak mode, the effect is similar to someone speaking down a long cardboard tube. This can, under certain conditions, be quite effective in improving the readability of the station. Therefore the use of Peak or Notch will depend on the listening conditions and type of signal being received and only by experience will you obtain the best results from your unit.

If you already have an audio filter in line with your speaker or headphones, i.e. the Maplin 'DXers Audio Processor Kit' (LK05F), it can be left connected. Under these conditions the effects of both units is compounded, so providing even more processing of the audio signal.

We hope that you will get as much pleasure out of building and using this project as we have, and that it will enhance your enjoyment of listening on the radio, free of some of the annoying QRM!

NOTCH FILTER PARTS LIST

RESISTORS: All 0.6W 1% Metal Film			
R1	6k8	1	(M6K8)
R2	1M	1	(M1M)
R3	27k	1	(M27K)
R4	18k	1	(M18K)
R5,6,12	4k7	3	(M4K7)
R7,27,28,29,30	10k	5	(M10K)
R8,10	2k2	2	(M2K2)
R9,20,38,39	1k8	4	(M1K8)
R11	10R 1W 5% Carbon Film	1	(C10R)
R13	22k	1	(M22K)
R14,15	47k	1	(M47K)
R16	8k2	1	(M8K2)
R17	2R7	1	(M2R7)
R18,19	100R	2	(M100R)
R21,25,26,31,32, 36,37,42,43,44	100k	10	(M100K)
R22,24,33,35	33k	4	(M33K)
R23,34	270k	2	(M270K)
R40	470k	1	(M470K)
R41	39k	1	(M39K)
R45	1k5	1	(M1K5)
R46	240k	1	(M240K)
R47	470R	1	(M470R)
R48	120k	1	(M120K)
RV1	10k Pot lin	1	(FW02C)
CAPACITORS			
C1,17,25	1μF 100V PC Electrolytic	3	(FF01B)
C2,5,12,18,20, 26,28,31,32	100nF Minidisc	9	(YR75S)
C3,4,6,8,9,21,22, 27,29,30	10μF 50V PC Electrolytic	10	(FF04E)
C7	220pF Ceramic	1	(WX60Q)
C10	47pF Ceramic	1	(WX52G)
C11	4μ7F 63V PC Electrolytic	1	(FF03D)
C13	150nF Polylayer	1	(WW43W)
C14,16	470μF 25V PC Electrolytic	2	(FF16S)
C16,19,23	150pF Ceramic	3	(WX58N)
C24	390pF Polystyrene 1%	1	(BX52G)
SEMICONDUCTORS			
D1,2	1N4148	2	(QL80B)
D3	1N4001	1	(QL73Q)
TR1	BC548	1	(QB73Q)
L1	Choke 10μH	1	(WH35Q)
IC1,3	LF353	2	(WQ31J)
IC2	LM380	1	(QH40T)
IC4	MF10CN	1	(QY35Q)
IC5	LF351	1	(WQ30H)
IC6	4046BE	1	(QW32K)

MISCELLANEOUS			
	Notch Filter PCB	1	(GD30H)
	LED Clip	4	(YY40T)
	DIL Socket 8-pin	3	(BL17T)
	DIL Socket 14-pin	1	(BL18U)
	DIL Socket 16-pin	1	(BL19V)
	DIL Socket 20-pin	1	(HQ77J)
SK2,3,4,5	Lever Terminal 4-way	1	(BW71N)
	Knob K7B	1	(YX02C)
	Spacer Threaded 6BA x ¼in.	1 Pkt	(FD10L)
	Bolt 6BA x 1in.	1 Pkt	(BF07H)
	Bolt 6BA x ½in.	1 Pkt	(BF06G)
	Nut 6BA	1 Pkt	(BF18U)
	Fuse Clips	2	(WH49D)
FS1	Fuse 500mA 20mm	1	(WR02C)
SK1	Power Socket 2.5mm	1	(FK06G)
S2,3,4	Latch Switch 2-pole	3	(FH67X)
S1	Latch Switch 4-pole	1	(FH68Y)
LD1,3,4	Red LED	3	(WL27E)
LD2	Green LED	1	(WL28F)
JK1	Jack Socket 3.5mm	1	(FK20W)
JK2	Jack Socket ¼in. standard	1	(FJ87U)
	Ribbon Cable 10-way	1 mtr	(XR06G)
	Minicon Ltch Hsg 2-way	4	(HB59P)
	Minicon Terminal	8	(YW25C)
	Shake 6BA	1 Pkt	(BF26D)
	Rct Latchbutton Black	1	(FH61R)
	Rct Latchbutton Grey	1	(FH62S)
	Rct Latchbutton Red	1	(FH63T)
	Rct Latchbutton White	1	(FH64U)
	Screening Can High	1	(FP57M)
	Screening Can Low	1	(FP58N)
OPTIONAL			
	Blue Case 212	1	(XY43W)
	Front Panel	1	(FP05F)
	Standard Power Plug 2.5mm	1	(HH62S)

A complete kit of all parts, excluding optional items, is available:
Order As LM16S (Notch Filter Kit) Price £29.95
The following are also available separately, but are not shown in the 1987 catalogue:
Notch Filter PCB **Order As GD30H Price £11.95**
Notch Filter Front Panel **Order As FP05F Price £2.95**
Screening Can High **Order As FP57M Price 68p**
Screening Can Low **Order As FP58N Price 60p**

WHICH RESISTOR?

by J.K. Hearfield

It is difficult to think of a circuit that doesn't include resistors. Over 10 billion of them are used each year in Europe alone. But like any other component, resistors have their limitations, and a careful designer will take them into account when deciding which resistor to choose for a particular application.

Preferred Values

Resistors are generally sold as preferred values in the range 10 ohms to 1 megohm, though some types are available in values down to 0.1 ohms and others up to 100 megohms.

The preferred values (10 ohms, 12 ohms, 15 ohms and so on) reflect the era when a tolerance of plus or minus 10% was standard and each decade of resistance could be covered with just 12 values, see Figure 1. Resistors today are commonly available at 1%, 2% and 5% tolerance in the E24 and E96 preferred value ranges, so called because they define respectively 24 and 96 values in each decade, see Table 1.

Other Values

Each preferred value in the E24 range is about 10% larger than the one below, so it is always possible to find a preferred value within 5% of the value you actually want. Sometimes though, a precise non-standard value is essential, and it then becomes necessary to make one up specially by putting two resistors in parallel (or series). Combining resistors in this way has an unexpected side benefit: when two resistors of 1% tolerance are connected in parallel (or series) the tolerance of the combination cannot be greater than 1%. Indeed, if each errs in the opposite direction by just the right amount, the net error can be zero!

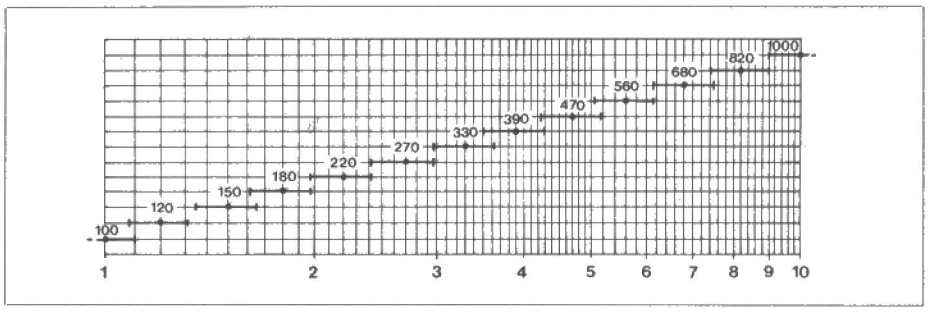


Figure 1. Twelve 10% resistors cover a decade.

Table 2 can be used to find the two preferred values which in parallel will yield the odd value you need.

Colour Code

Manufacturers usually make resistors with a series of coloured bands or sometimes dots. The colour of each band and its position in the sequence defines the component's value and tolerance, as shown in Figure 2.

Components with three, four or six bands can be identified easily enough, but there are unfortunately two separate and different standards for labelling five-band resistors:

	Band 1	Band 2	Band 3
MAPLIN:	First digit	Second digit	Multiplier
OTHER:	First digit	Second digit	Third digit
	Band 4	Band 5	
MAPLIN:	Tolerance	Tempco	
OTHER:	Multiplier	Tolerance	

This may cause confusion if your resistors get mixed up - for example, a Maplin 120 ohm 1% E24 metal film type (which has the colour code Brown, Red, Brown, Brown, Red) could be read

E24 Preferred Values.

1.0	1.1	1.2	1.3	1.5	1.6
1.8	2.0	2.2	2.4	2.7	3.0
3.3	3.6	3.9	4.3	4.7	5.1
5.6	6.2	6.8	7.6	8.2	9.1

E96 Preferred Values.

1.00	1.02	1.05	1.07	1.10	1.13	1.15	1.18
1.21	1.24	1.27	1.30	1.33	1.37	1.40	1.43
1.47	1.50	1.54	1.58	1.62	1.65	1.69	1.74
1.78	1.82	1.87	1.91	1.96	2.00	2.05	2.10
2.15	2.21	2.26	2.32	2.37	2.43	2.49	2.55
2.61	2.67	2.74	2.80	2.87	2.94	3.01	3.09
3.16	3.24	3.32	3.40	3.48	3.57	3.65	3.74
3.83	3.92	4.02	4.12	4.22	4.32	4.42	4.53
4.64	4.75	4.87	4.99	5.11	5.23	5.36	5.49
5.62	5.76	5.90	6.04	6.19	6.34	6.49	6.65
6.81	6.98	7.15	7.32	7.50	7.68	7.87	8.06
8.25	8.45	8.66	8.87	9.09	9.31	9.53	9.76

Table 1. E24 and E96 values.

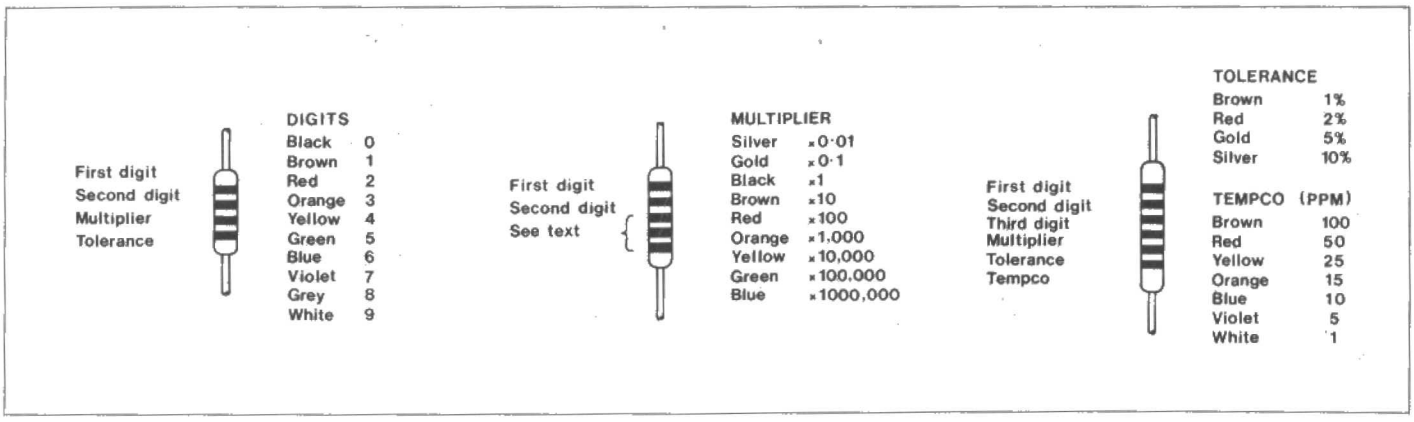


Figure 2. Resistor colour codes.

wrongly as a 2% E96 resistor of 1210 ohms.

Some people use a mnemonic to remember the colour code. The one I was taught is unforgettable but obscene, but it's easy to invent your own. Here's one:

Black Birds Run Over Your Garden Biting Visible Grey Worms.

Temperature Effects

A resistor is a device for turning electrical energy into heat, some run quite hot enough to burn an incautious finger. As a resistor warms up, its value changes slightly; the hotter it gets, the bigger the change. Its resistance also starts to drift permanently away from the nominal value. To keep both these effects in check, manufacturers define a maximum safe operating temperature (the hot-spot temperature) for their components, and to make life easier for designers, it is specified in terms of the maximum amount of power the component can safely dissipate.

The temperature coefficient of a resistor - usually abbreviated to tempco - is quite small, typically 10 to 100 parts per million per degree (C). To put this in perspective, the resistance of a component having a tempco of 50ppm will differ from its nominal value by about 0.2% at 60°C. Tempco is sometimes marked on the component itself as an extra coloured band.

If long-term stability is important, the resistor's operating temperature sould be kept as low as possible. A typical $^1/_2$ watt metal film type dissipating 200mW at 70°C may drift by 0.5% in the course of a year; at 40°C the drift would be only half as much.

Power Dissipation

When a resistor dissipates power, its surface temperature rises until it can get rid of the heat as fast as it is being generated. The surface temperature is then given by:

(Power dissipated x Thermal resistance) + (Ambient temperature)

Thermal resistance is a measure of how efficiently a component warms the surrounding air; it has nothing to do with ohmic resistance. Typical figures are:

Small $^1/_8$ watt type: 300°C/watt
Ordinary $^1/_2$ watt type: 200°C/watt
1 watt type: 100°C/watt
6 watt type: 50°C/watt

As an example of what this means in practice, suppose a resistor must dissipate 400mW in an ambient temperature of 45°C. Either a $^1/_2$ watt or a 1 watt component could be used. The $^1/_2$ watt type would run at a surface temperature of:

(0.4 x 200) + 45 = 125°C

Whilst the physically larger 1 watt type would run rather cooler at:

(0.4 x 100) + 45 = 85°C

The thermal resistance must be low if the resistor is to dissipate large amounts of power. The obvious way to achieve this is by arranging that the

Table 2

	100	110	120	150	160	180	200	220	240	270	300	330	360	390	430	470	510	560	620	680	750	820	910
100	50	52.4	54.5	60	61.5	64.3	66.7	68.8	70.6	73	75	76.7	78.3	79.6	81.1	82.5	83.6	84.8	86.1	87.2	88.2	89.1	90.1
110	52.4	55	57.4	63.5	65.2	68.3	71	73.3	75.4	78.2	80.5	82.5	84.3	85.8	87.6	89.1	90.5	91.9	93.4	94.7	95.9	97	98.1
120	54.5	57.4	60	66.7	68.6	72	75	77.6	80	83.1	85.7	88	90	91.8	93.8	95.6	97.1	98.8	101	102	103	105	106
150	60	63.5	66.7	75	77.4	81.8	85.7	89.2	92.3	96.4	100	103	106	108	111	114	116	118	121	123	125	127	129
160	61.5	65.2	68.6	77.4	80	84.7	88.9	92.6	96	100	104	108	111	113	117	119	122	124	127	130	132	134	136
180	64.3	68.3	72	81.8	84.7	90	94.7	99	103	108	113	116	120	123	127	130	133	136	140	142	145	148	150
200	66.7	71	75	85.7	88.9	94.7	100	105	109	115	120	125	129	132	137	140	144	147	151	155	158	161	164
220	68.8	73.3	77.6	89.2	92.6	99	105	110	115	121	127	132	137	141	146	150	154	158	162	166	170	173	177
240	70.6	75.4	80	92.3	96	103	109	115	120	127	133	139	144	149	154	159	163	168	173	177	182	186	190
270	73	78.2	83.1	96.4	100	108	115	121	127	135	142	149	154	160	166	171	177	182	188	193	199	203	208
300	75	80.5	85.7	100	104	113	120	127	133	142	150	157	164	170	177	183	189	195	202	208	214	220	226
330	76.7	82.5	88	103	108	116	125	132	139	149	157	165	172	179	187	194	200	208	215	222	229	235	242
360	78.3	84.3	90	106	111	120	129	137	144	154	164	172	180	187	196	204	211	219	228	235	243	250	258
390	79.6	85.8	91.8	108	113	123	132	141	149	160	170	179	187	195	205	213	221	230	239	248	257	264	273
430	81.1	87.6	93.8	111	117	127	137	146	154	166	177	187	196	205	215	225	233	243	254	263	273	282	292
470	82.5	89.1	95.6	114	119	130	140	150	159	171	183	194	204	213	225	235	245	256	267	278	289	299	310
510	83.6	90.5	97.1	116	122	133	144	154	163	177	189	200	211	221	233	245	255	267	280	291	304	314	327
560	84.8	91.9	98.8	118	124	136	147	158	168	182	195	208	219	230	243	256	267	280	294	307	321	333	347
620	86.1	93.4	101	121	127	140	151	162	173	188	202	215	228	239	254	267	280	294	310	324	339	353	369
680	87.2	94.7	102	123	130	142	155	166	177	193	208	222	235	248	263	278	291	307	324	340	357	372	389
750	88.2	95.9	103	125	132	145	158	170	182	199	214	229	243	257	273	289	304	321	339	357	375	392	411
820	89.1	97	105	127	134	148	161	173	186	203	220	235	250	264	282	299	314	333	353	372	392	410	431
910	90.1	98.1	106	129	136	150	164	177	190	208	226	242	258	273	292	310	327	347	369	389	411	431	455
1000	90.9	99.1	107	130	138	153	167	180	194	213	231	248	265	281	301	320	338	359	383	405	429	451	476
1100	91.7	100	108	132	140	155	169	183	197	217	236	254	271	288	309	329	348	371	397	420	446	470	498
1200	92.3	101	109	133	141	157	171	186	200	220	240	259	277	294	317	338	358	382	409	434	462	487	518
1500	93.8	102	111	136	145	161	176	192	207	229	250	270	290	310	334	358	381	408	439	468	500	530	566
1800	94.7	104	113	138	147	164	180	196	212	235	257	279	300	321	347	373	397	427	461	494	529	563	604
2000	95.2	104	114	142	152	171	190	209	227	256	284	313	341	370	408	445	483	531	588	645	711	777	863

Table 2. Preferred values in parallel.

resistor has a large surface area; this is why high-power resistors are physically large. Another method is to package the component inside its own heatsink, which can in turn be bolted to a bigger heatsink to reduce the operating temperature still further.

Figure 3 illustrates how the physical sizes of the resistor types in the Maplin range depend on their power ratings.

Voltage Limitations

Resistors can be damaged by excessive voltage stress even though their power rating is not exceeded. If the voltage across the resistor is ever likely to exceed about 250 volts, a metal glaze type may be the best solution.

Since the power dissipated by a resistor depends on the square of the voltage across it, the voltage and power restrictions can be combined into a single graph. Figure 4 shows the 'safe operating area' for each resistor type in the Maplin range.

Frequency Effects

All components possess some stray inductance (in their leads) and some stray capacitance. Figure 5 shows a model of a real resistor.

The inductance of a length of straight wire is about 15nH/cm. Taking into account the stray inductance in the PCB tracks and in the leads themselves the total series inductance might amount to 20nH. This would make a 10 ohm resistor begin to look inductive at around 100MHz; for higher-value resistors, the effects would appear at correspondingly higher frequencies. A stray capacitance of 1pF would begin to affect the performance of a 100k ohm resistor at frequencies around 1MHz; smaller resistors would be affected at correspondingly higher frequencies.

Noise

Resistors generate electrical noise. Their Johnson noise can be calculated from the expression:

$v = SQR (4 k T R B)$

Where v is the noise voltage, k is Boltzmann's constant (1.37E-23), T is the absolute temperature, R is the resistance (ohms) and B is the bandwidth. The formula predicts that a 10k ohm resistor will generate about 1μV of noise in a 10kHz bandwidth at room temperature. In practice, most resistors are noisier than this, especially when they are carrying a *direct current*. The extra noise they produce - which is concentrated at low frequencies - depends linearly on the applied voltage and is usually quoted in μV/V. Carbon film types are the noisiest; metal film and wirewound types are the quietest.

Resistor Types

Carbon composition resistors were at one time the most popular and widely-used types. They are cheap, though they have a large positive tempco, especially at extreme temperatures, and a high noise figure. They have now been largely superseded by other types.

The main advantage of carbon film resistors is their cheapness, though their negative temperature coefficient (around −300ppm) makes them specially useful in some applications. Their noise figure may be as high as 5μV/V. They are typically sold with 5% tolerance.

High stability and low noise (0.1μV/V) make metal film resistors the preferred choice for most designs. Their tempco is low: typcially +50 ppm. They are widely available at 1% and 2% tolerance and they are almost the cheapest resistor you can buy.

Metal glaze resistors tend to be used in applications needing good surge and overload capability. Their tempco is slightly higher than that of metal film types - typically +100ppm. They are available at 1% and 2% tolerances, though they tend to cost around twice as much as their metal film equivalents.

Metal oxide resistors offer good precision with above-average power-handling capability in the range from tens of ohms to hundreds of kilohms. Their tempco is quite high - typically +200ppm. They cost about the same as equivalent wire wound resistors.

Wire wound resistors find their chief use where several watts of power must be dissipated or where extreme stability is vital. They are available at 5% and 10% tolerance in values from below 0.1 ohm to a few tens of kilohms with power ratings up to 100 watts. Their chief disadvantage is their poor high-frequency performance.

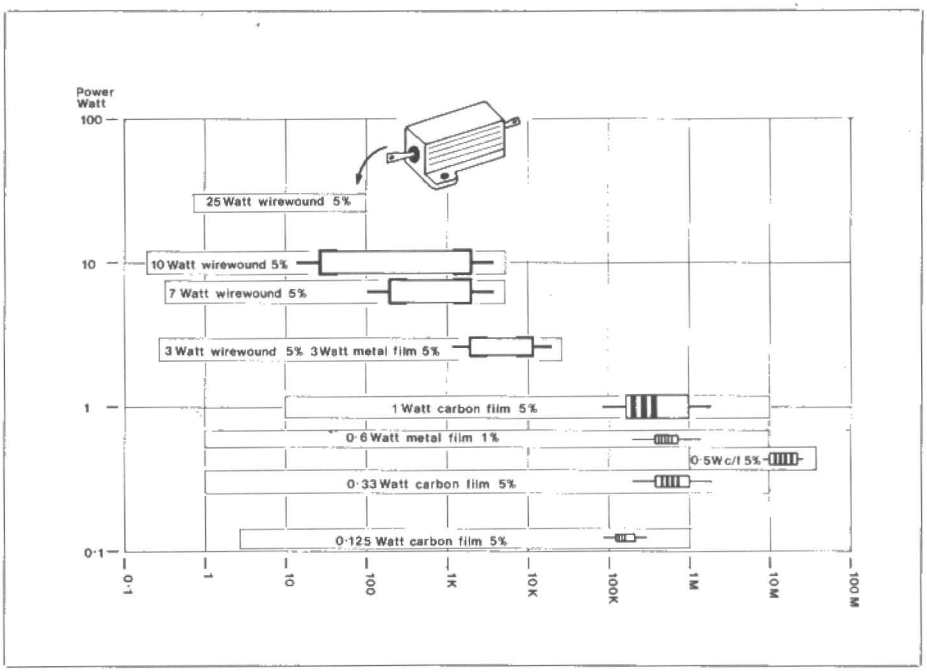


Figure 3. Power ratings of resistor types.

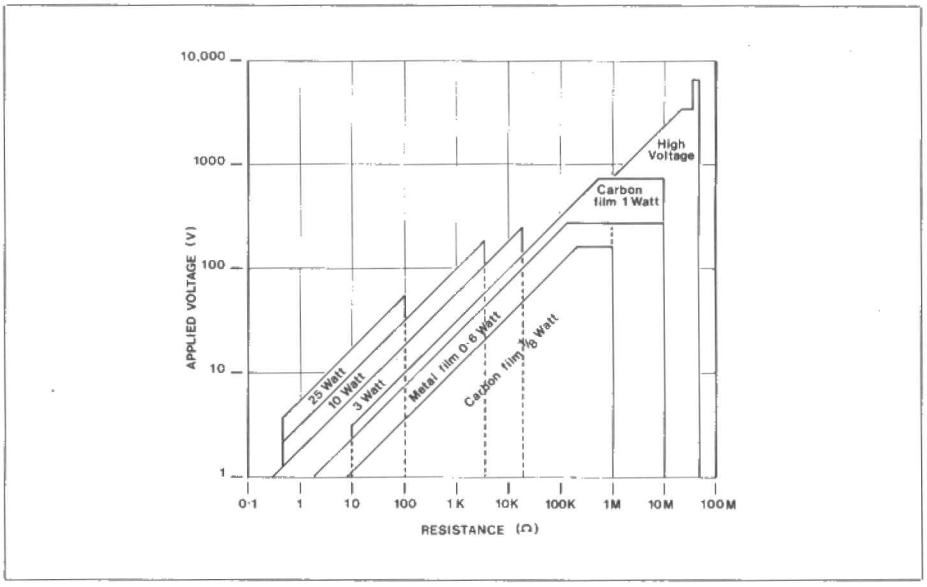


Figure 4. Safe operating area for resistors.

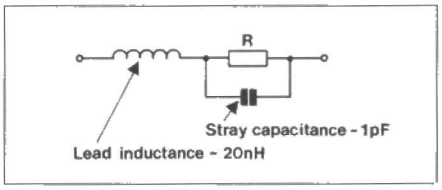


Figure 5. Model of a real resistor.

PUBLIC ADDRESS SYSTEM

★ **Total 16 Watts RMS Output**
★ **Maximum of 10 x 8Ω Speakers can be Driven**
★ **Supplied from 12V Car Battery**

by Dave Goodman

Introducing a single input (mono) amplification system for processing speech from a microphone and driving a maximum of ten 8Ω speakers with a total power output of 16 watts r.m.s. (30 Watts pk).

A 12V accumulator type car battery is used as a power source, making the unit portable enough for use at outdoor events, such as fetes and sports days or wherever public addressing is required.

Public Addressing

Although the human ear is capable of processing sound over most of the audio bandwidth, speech information is at its greatest intelligibility between 300Hz and 3kHz. Very low and high frequency harmonics in speech areas do not convey much information content and for PA use, can be removed. The narrow, bandpass area remaining is amplified and out of band signals will not then excessively distort the waveform.

Rather than use one or two loudspeakers at very high power levels to communicate at outdoor events, it is far more practical to distribute several speakers around the central area to be covered. Of course, power output levels from each speaker will be reduced accordingly, but the total will be distributed evenly and not just in one place. Without power bridging techniques or inverter power supplies being utilised, only a relatively low level (10 - 20W r.m.s.) of power output can be generated from a 12V supply. This project uses two amplifiers each capable of driving loads down to 1.6Ω impedance or five eight ohm speakers connected in parallel. Approximately 8 watts (16 watts pk) can be delivered to each of the two associated outputs achieving a combined total of 16 watts (30W pk) from ten speakers.

SPECIFICATION

Supply Voltage	6V to 17V DC
Min Supply Current	100mA
Max Supply Current	2.25A
Rated Power Output	8W RMS per channel (16W pk) Both channels running = 16W RMS total @ 1kHz
Input Sensitivity	2mV (0.002V) continously variable for rated output
Load Impedance	1.6Ω max to 8Ω min
Power Response	680Hz to 2kHz

Standard ¼in. Jack Socket Microphone Input. Two ¼in. Jack Sockets for Speaker Output. Input Sensitivity Preset and Volume Control.

Circuit Description

In Figure 1, TR1 and TR2 form a simple, common emitter pre-amp stage with a gain of 200 set by R4. The input signal from JK1 is preset by RV1 and can be adjusted for input levels between 1mV and 1V. High impedance microphones (47kΩ) generate signal levels around 100mV, but low impedance microphones (200-600Ω) produce a very low output signal, often less than 1mV. RV1 is therefore preset to suit the microphone in use. R1 is referenced as a S.O.T. (Select On Test) resistor which can be fitted to terminate low-Z mic cables, if required. In this instance R1 should have a resistance of 680Ω.

TR3 buffers the pre-amp output and offers a low impedance drive to the limiter stage C4, D1 and D2. With a low supply rail and large dynamic range microphones, it becomes quite easy to overdrive the power amp output stage and clipped waveforms produce excessive distortion! D1 and D2 are biased to limit large transient signals, but even so, harmonics are generated which must be low pass filtered. TR4 forms the active component in the filter stage and attenuates signals above 1kHz at 6dB per octave. The combined responses of the limiter and filter stage produce the speech, Band Pass characteristic shown in Figure 2.

Volume control RV2 sets the signal drive level to power amp IC1. This device is configured as a 'Dual Bus' amplifier, having two separate amplifiers driven from the same source signal and capacitively coupled to the output jacks.

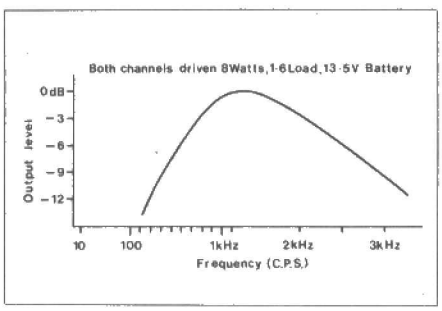


Figure 2. Power Response.

Construction

To begin the pcb assembly, three links should be inserted; one below R15, one below C10 and one below C23. (Refer to Figure 3 and the Constructors Guide.) Identify and insert resistors R2 to R21 from the pcb 'component side' and both wirewound resistors R22, 23. Insert D1, D2 and solder these components. Cut off spare wire ends and mount the capacitors and RV1. Ensure electrolytic types are fitted correctly and insert polylayer types with the minimum of force, to avoid breaking off their leads. Solder these components to prevent them falling out later on and fit the four transistors and terminal block. Orientate TB1 so that the terminals face towards the back edge of the pcb. Mount the three

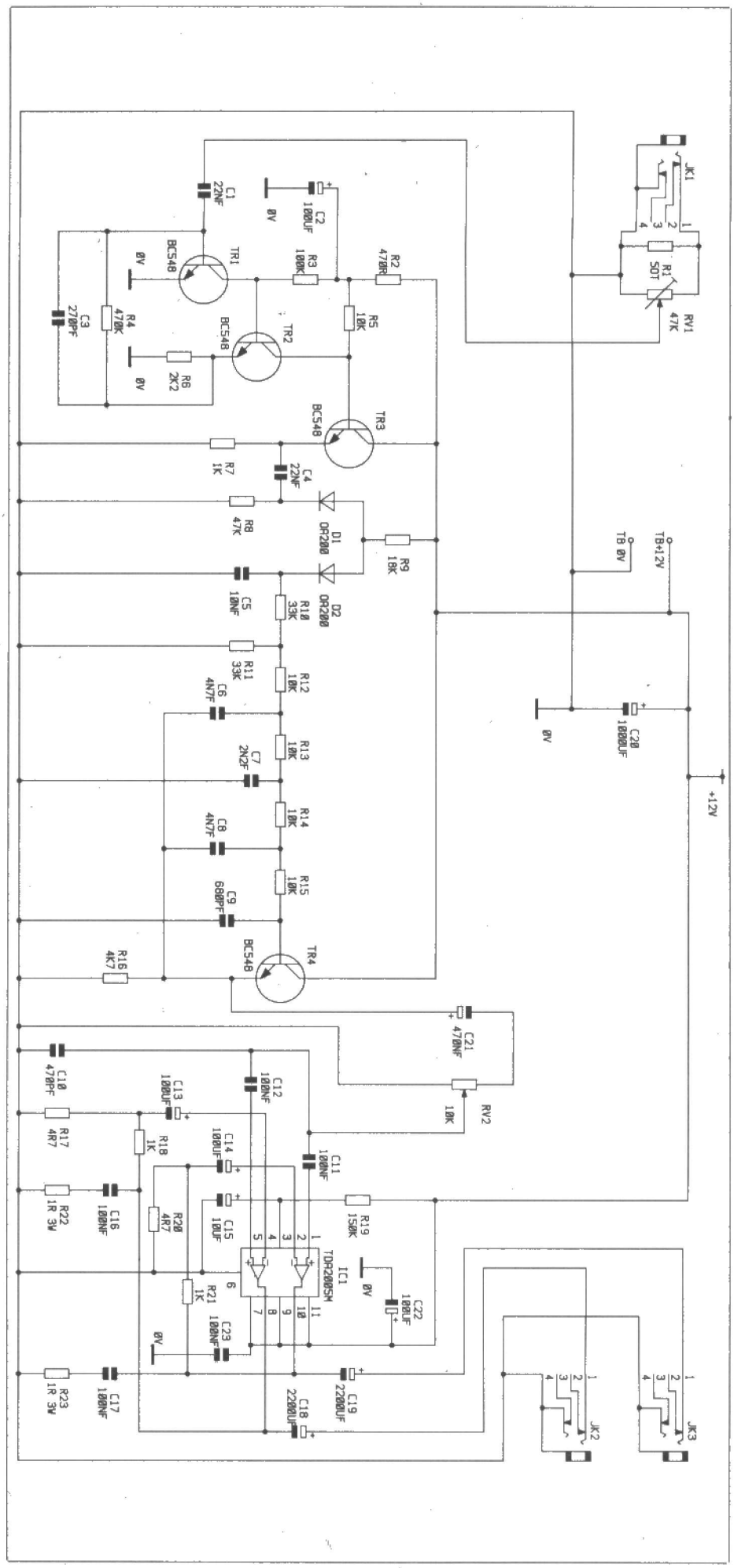
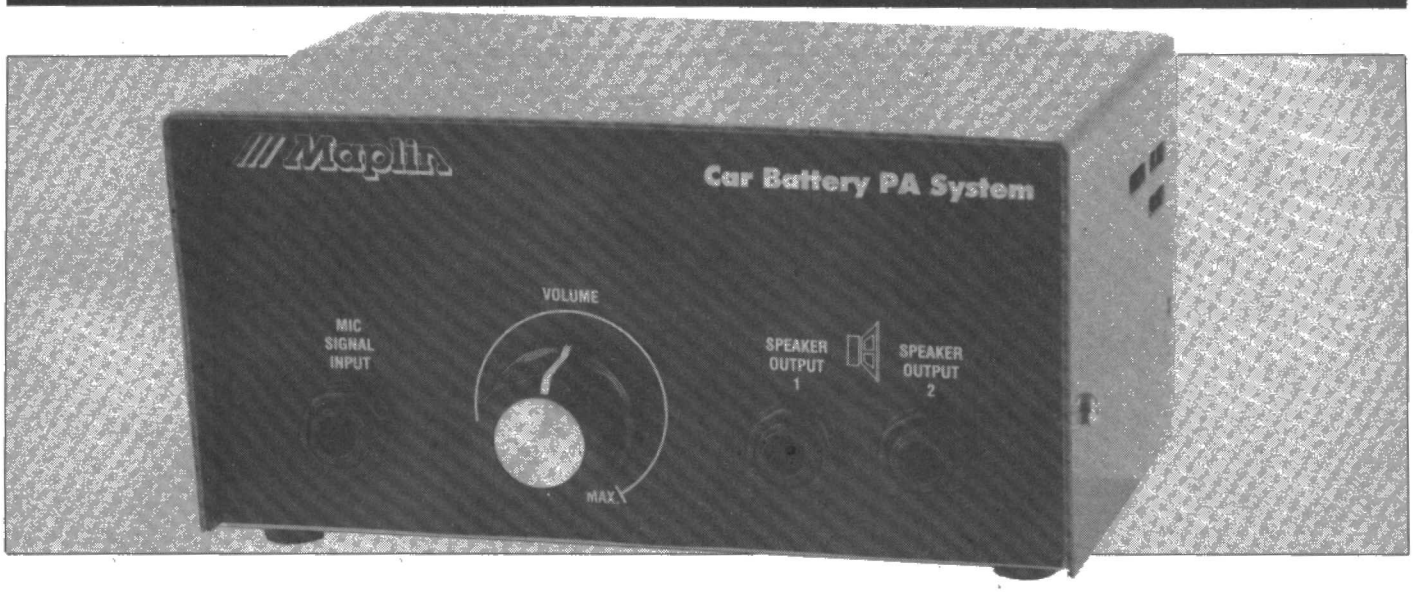


Figure 1. Circuit Diagram.



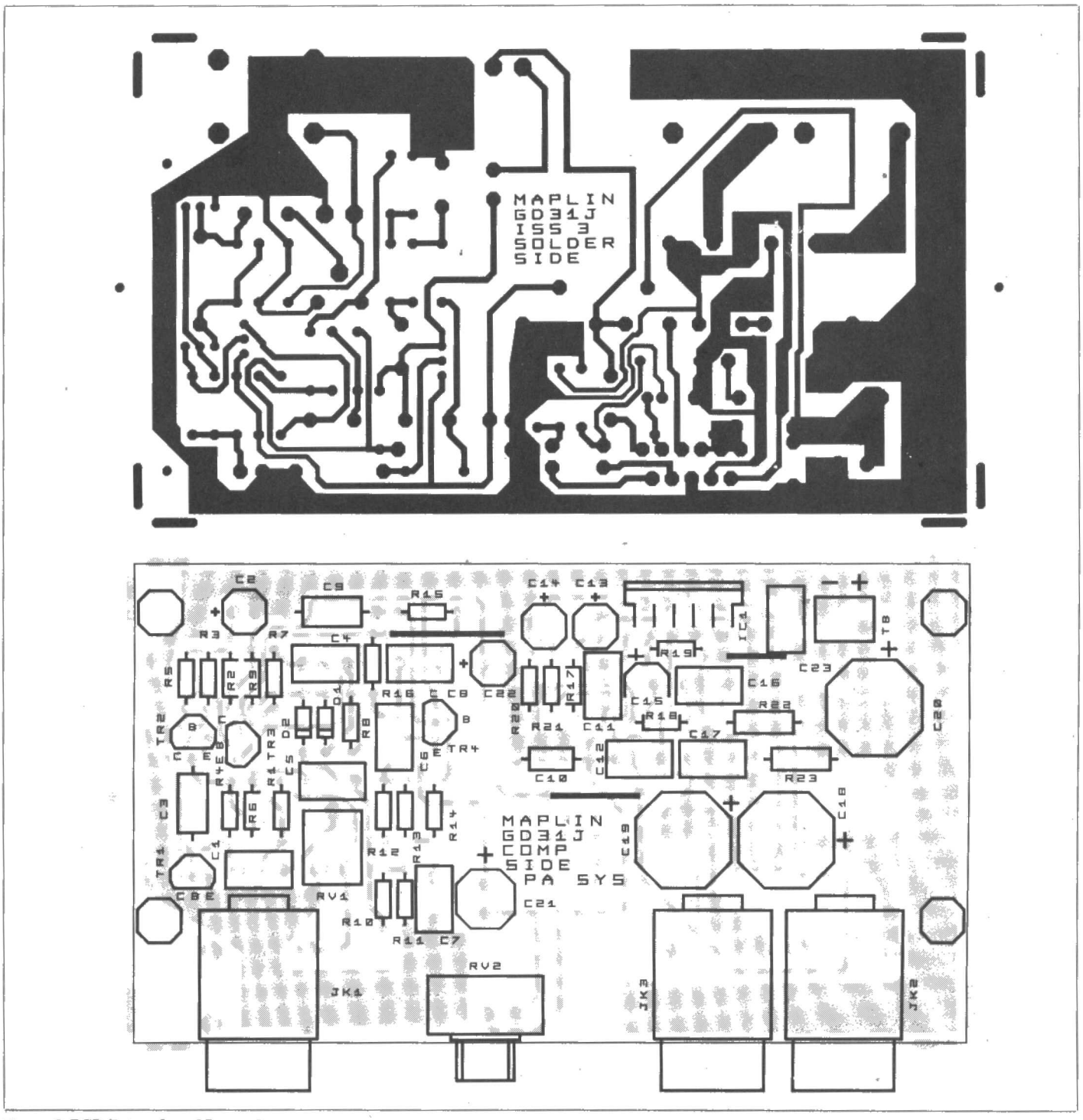
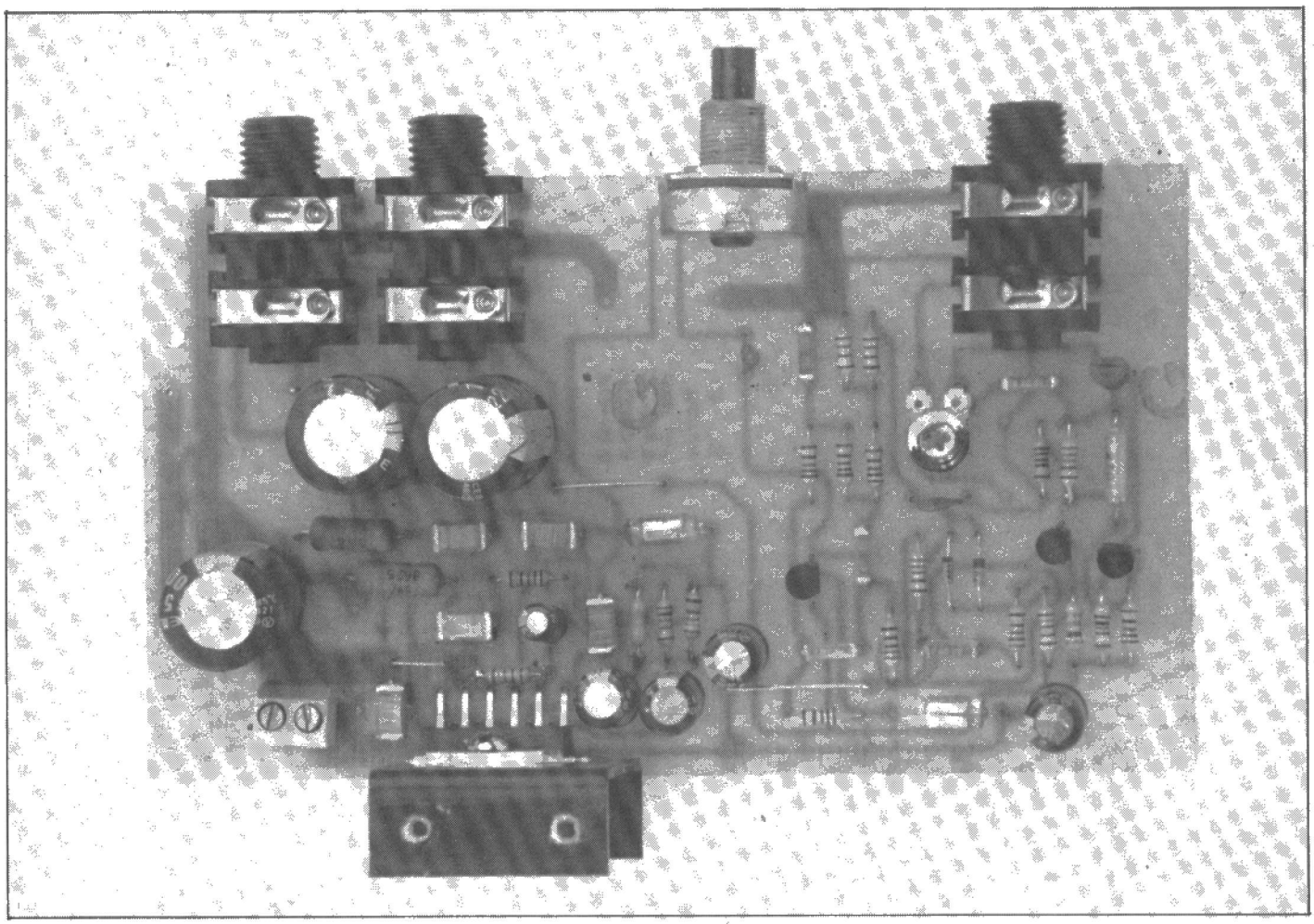


Figure 3. PCB Artwork and Legend.



jack sockets and RV2. Solder the remaining components and after removing excess wire ends, clean the track area for inspection.

Before mounting IC1, refer to Figure 4. The black heatsink mounting bracket has to be drilled before fitting and two 2mm holes are required. Carefully insert a No.4 self-tap screw into each of these holes and cut a thread. Make sure the screws can be tightened up and remove before continuing with the assembly. The bracket in Figure 4 is shown with both ends facing toward the front and *must* be orientated with the four holes positioned as shown. Spread a layer of thermal compound onto the metal tab of IC1 and fit onto the lower-middle hole in the bracket, using a 4BA x ¼in. bolt and nut as Figure 5. Fit this assembly onto the pcb (Figure 6) so that all eleven pins of IC1 are inserted. Then position IC1 by adjusting the depth of all 11 pins such that a measurement of 18mm from the centre of the 4BA bolt and the surface of the pcb is achieved.

Solder IC1 in position ensuring the assembly sits at right angles to the board.

Assembly

Cut the plastic spindle of RV1 approximately 7mm from the threaded bush, and gently push RV1 back towards the board. Next place a 6BA x ½in. bolt into each of the two holes in the box base. Use the two holes nearest to the box back panel, and screw a hexagonal threaded spacer onto each bolt, from the inside. Insert the completed module (Figure 7) front edge down so that RV1 enters its hole position first, followed by the three jack socket bushes. Slide the module into position over both 6BA bolts and spacers,

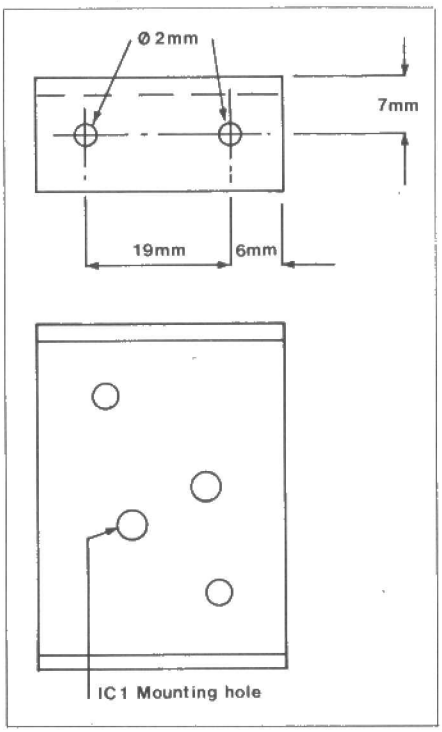


Figure 4. Heatsink Bracket Drilling.

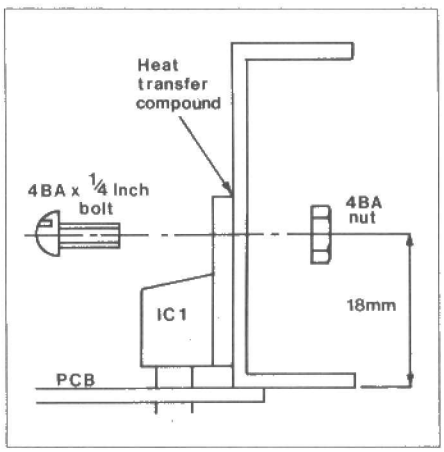


Figure 5. Mounting IC1.

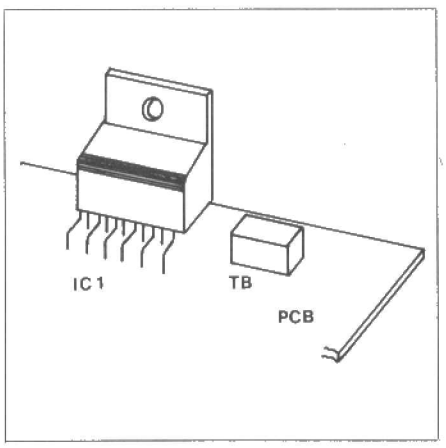
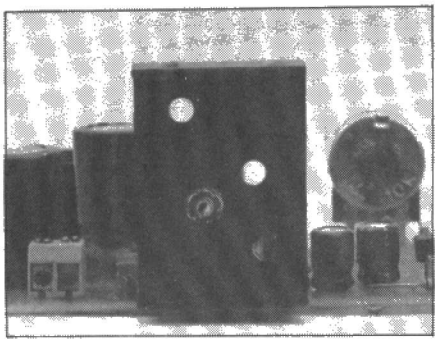


Figure 6. Heatsink Sub-assembly.



and fit a locking hexagonal spacer to each bolt to hold the board in position.

At this stage, wire two cables into the terminal block, for connections to the battery supply. Once the heatsink is fitted, the terminal block is *not accessible* so this should be done now. Use 6 Amp capacity cable and keep the cable length as short as possible to the battery.

Heatsink Assembly

Spread a thin layer of thermal compound over the previously drilled end of IC1 mounting bracket and also along the length of the drilled edge on the aluminum heatsink panel (Figure 8). Align two holes in the panel with both holes on the mounting bracket and insert two No.4 self-tapping screws. Fit the heatsink panel to the box back panel with two No.4 self-tapping screws and tighten-up all four screws.

Finally, stick the self-adhesive legend onto the box front panel and fit all four lock nuts to the jack sockets and bush of RV1. Loosen the locking screw in the volume control knob and fit this too. The box cover can be fitted with the sloping edges to the front and positioned with four cross-head screws supplied with the case. Peel off each of the self-adhesive foot pads and stick these under the box base in each of the four corners.

Using the PA Amplifier

Hi-Fi speaker systems are not really suitable for use in this application, but mid-range speakers or horns are ideal. Various horns, suitable for use with the amp are to be found in the speakers section of the Maplin 1987 'Buyers Guide' catalogue (and shown as optional items in the Parts List), although a suitable, waterproof mylar cone speaker could also be used, especially if you are intending to fit all ten 8Ω speakers. Figure 9 shows external connections to the amp, especially the speaker arrangements. The maximum number of 8Ω speakers connected in parallel on one output is five, giving a total load impedance of 1.6Ω. The total power developed in this load is 8 to 10 Watts r.m.s. or approximately 2 Watts per speaker. This applies to both output channels for a total of ten speakers at 20 Watts.

For in-car use, it is usual to have 4 horn speakers mounted cardinally onto a luggage rack fitted onto the car roof. If using 8Ω horns, connect two in parallel to

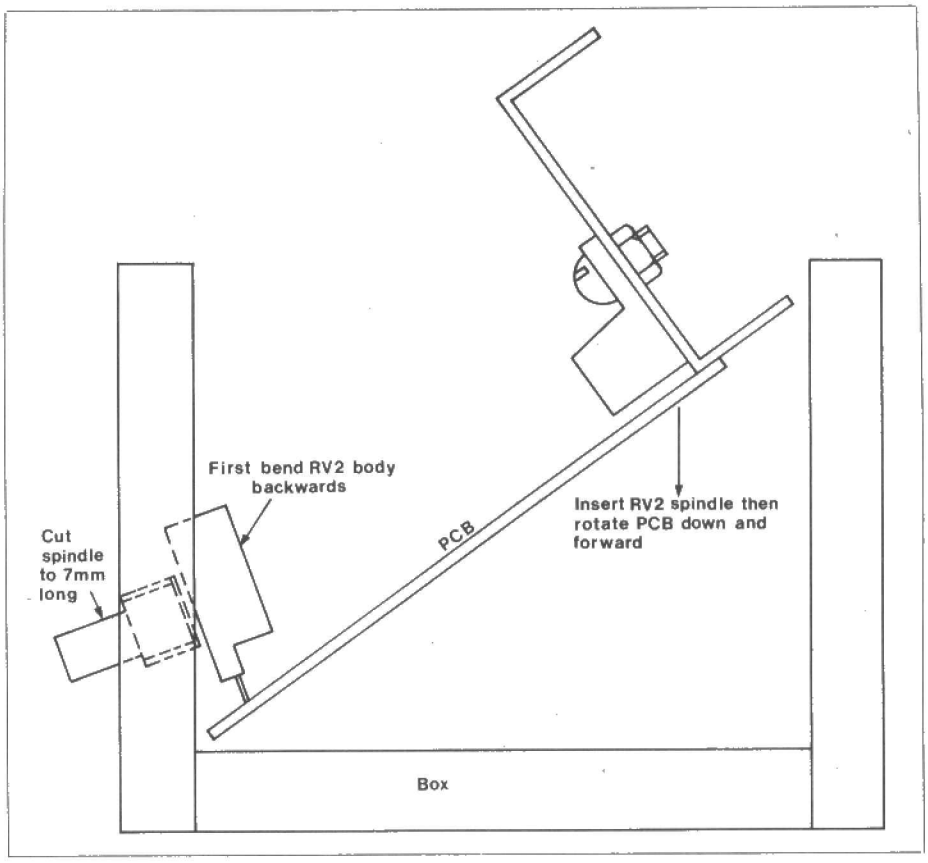
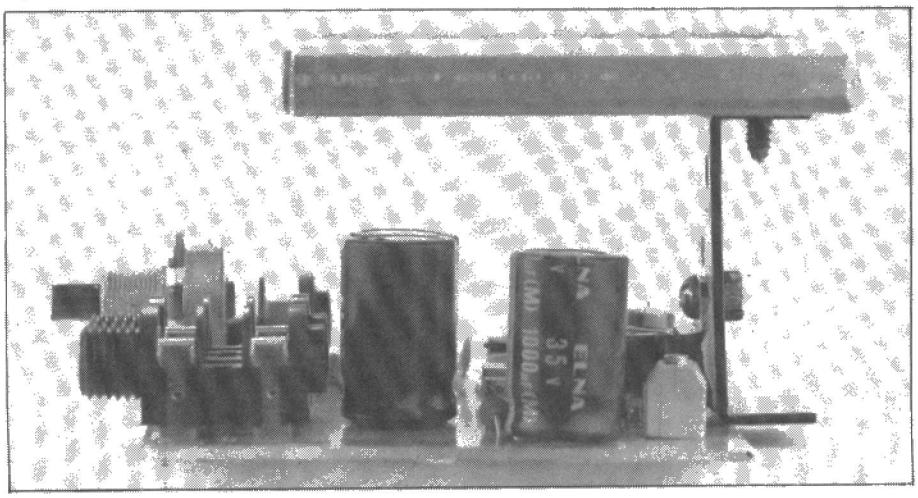


Figure 7. Fitting PCB into Case.



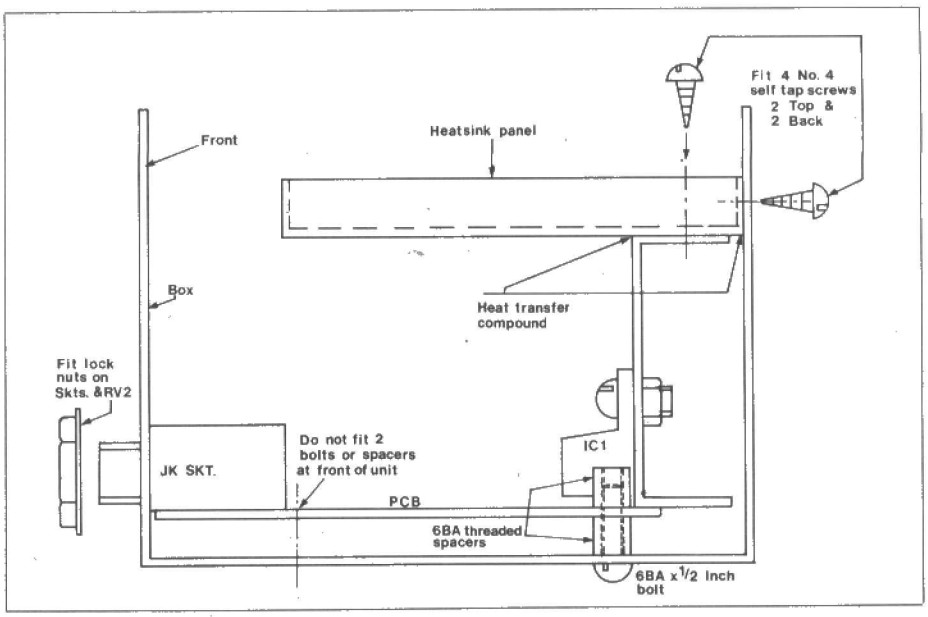

Fit 4 No. 4 self tap screws 2 Top & 2 Back
Front
Heatsink panel
Box
Heat transfer compound
Fit lock nuts on Skts. &RV2
JK SKT.
Do not fit 2 bolts or spacers at front of unit
IC1
PCB
6BA threaded spacers
6BA x 1/2 inch bolt

Figure 8. Final Assembly.

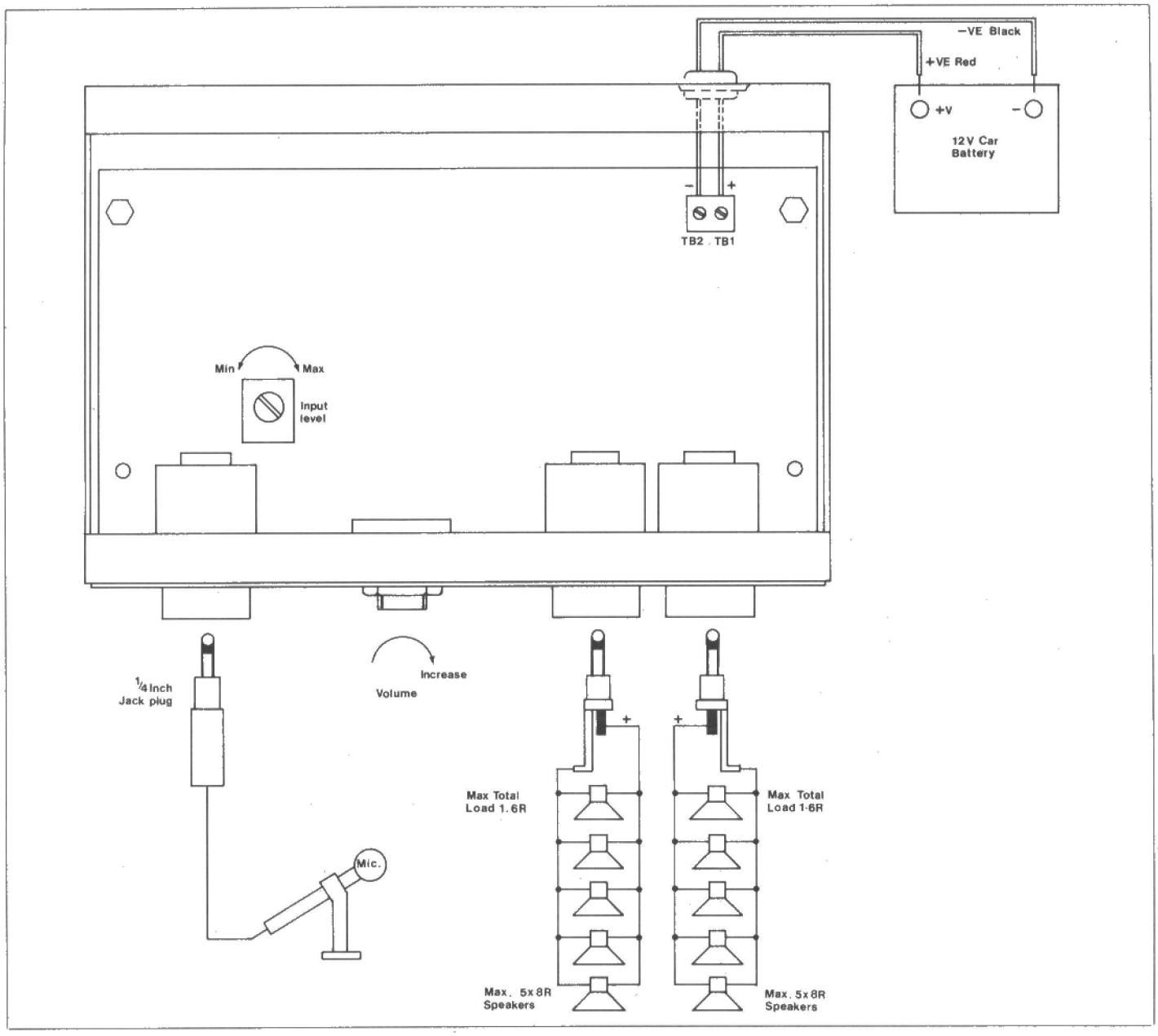


Figure 9. External Connections.

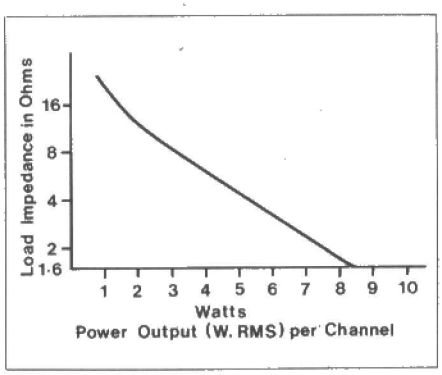
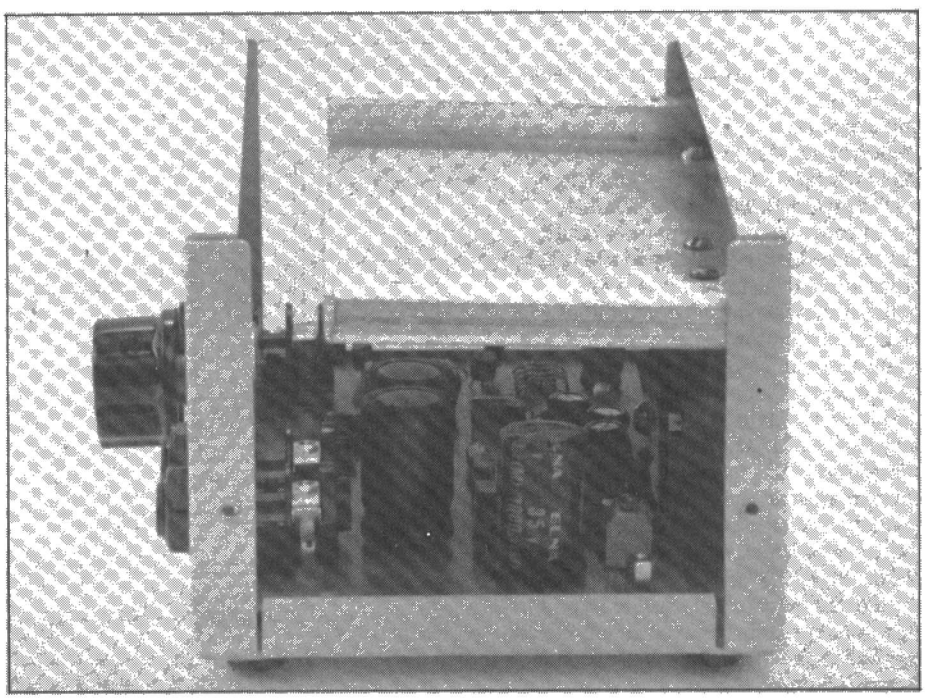


Figure 10. Power Output Levels.

channel 1 output, and the other two in parallel to channel 2 output. (Total load of 4Ω per channel). The graph in Figure 10 shows power output levels at various speaker load impedances. So for the car arrangement, approximately 10 Watts total power output is possible.

It should be noted that the amplifiers' response is tailored for speech and music signals will be reproduced with poor low and high frequency response!



PUBLIC ADDRESS SYSTEM PARTS LIST

RESISTORS: All 0.6W 1% Metal Film			
R1	Select on Test	1	See Text
R2	470R	1	(M470R)
R3	100k	1	(M100K)
R4	470k	1	(M470K)
R5,12,13,14,15	10k	5	(M10K)
R6	2k2	1	(M2K2)
R7,18,21	1k	3	(M1K)
R8	47k	1	(M47K)
R9	18k	1	(M18K)
R10,11	33k	2	(M33K)
R16	4k7	1	(M4K7)
R17,20	4R7	2	(M4R7)
R19	150k	1	(M150K)
R22,23	1R 3W W/Wound	2	(W1R)
RV1	47k Hor S/Min Preset	1	(WR60Q)
RV2	10k Pot Log	1	(FW22Y)
CAPACITORS			
C1,4	22nF Polylayer	2	(WW33L)
C2,13,14,22	100µF 25V PC Electrolytic	4	(FF11M)
C3	270pF 1% Polystyrene	1	(BX50E)
C5	10nF Polylayer	1	(WW29G)
C6,8	4n7F Polylayer	2	(WW26D)
C7	2n2F Polylayer	1	(WW24B)
C9	680pF Polysytrene	1	(BX34M)
C10	470pF Polystyrene	1	(BX32K)
C11,12,16,17,23	100nF Polylayer	5	(WW41U)
C15	10µF 50V PC Electrolytic	1	(FF04E)
C18,19	2200µF 16V PC Electrolytic	2	(FF60Q)
C20	1000µF 35V PC Electrolytic	1	(FF18U)
C21	470nF 35V Tantalum	1	(WW58N)
SEMICONDUCTORS			
TR1,2,3,4	BC548	4	(QB73Q)
D1,2	OA200	2	(QH74R)
IC1	TDA2005M	1	(YY70M)
MISCELLANEOUS			
JK1,2,3	Mono PCB ¼in. Jack Skt	3	(FJ00A)
TB	2-way PC Terminal	1	(FT38R)
	Knob K7B	1	(YX02C)
	Box Drilled	1	(YP12N)
	PA Board	1	(GD31J)
	Self Tap No.4 x ⅛in.	1 Pkt	(BF66W)
	Bolt 6BA x ½in.	1 Pkt	(BF06G)
	Bolt 4BA x ¼in.	1 Pkt	(BF02C)
	Nut 4BA	1 Pkt	(BF17T)
	Threaded Spacer 6BA x ¼in.	1 Pkt	(FD10L)
	Heatsink Basic TO3 Fin	1	(FG50E)
	Silicone Grease Tube	1 Tube	(HQ00A)
OPTIONAL			
	4in. Mylar Speaker 8R/15W	1	(YN03D)
	136mm Dia Horn 8R/10W	1	(XQ73Q)
	200 x 106mm Sq Horn 8R/30W	1	(XG97F)
	Jack Plug	2	(HF87U)
	Wire 3202 Black	As Rq	(XR32K)
	Wire 3202 Red	As Rq	(XR36P)
	Battery Clips	2	(HF26D)
	PA Front Panel	1	(FP59P)

A complete kit of all parts, excluding Optional items and R1, is available:
Order As LM17T (PA Kit) Price £19.95
The following are also available separately, but are not shown in the 1987 catalogue:
PA Board **Order As GD31J Price £2.95**
Pre-drilled Case **Order As YP12N Price £5.95**
PA Front Panel **Order As FP59P Price £2.95**

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A magnificent reproduction of a calendar that was first printed for the year 1987, when the dates in that year fell on the same days as they do next year (2387). The calendar features six superb pictures which appeared on the covers of the Maplin catalogue some 400 years ago, from 1979 to 1986, reproduced in full colour on genuine glossy art paper (not multiple image synthetics) with two months to each picture. The scenes offer a fascinating insight into the artists' ideas of how the future might have been. At that time, the calendar was sold for what now seems to be an unbelievably low price of just £3.50!! That's 'Sterling Earth Pounds', of course! And just for you, we are offering this reproduction calendar now for that same ridiculous original price (equivalent to 1.7170 intergalactic tokens). Be sure to order your copy now while stocks last! Overall size: approx. 37cm x 48cm.

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Watts in an Amplifier?

by Dave Goodman Part 1

Musicpower, Watts, RMS, Power and Decibels, are terms that appear to have mystical qualities to a great many people who wish to get involved with amplification projects! This series will attempt to unravel the mysteries and in addition describe amplifier projects from the range of kits available in the Maplin catalogue.

Power Amplifiers

The job of a power amplifier is to increase the level of signals delivered by previous pre-amplifier or control stages and make them suitable for driving loudspeakers, see Figure 1. For maximum power output delivery, a 'standard' signal level of 0.775V (775mV) or 0dB is usually applied to the amplifier input. This signal is then increased in level by some 30 to 50 times and reproduced by the loudspeaker at the appropriate sound level. Many demands are placed on amplifiers in the course of their duties as they must be able to accurately reproduce complex signal waveforms with the minimum of change and distortion. They must be capable of high current delivery into low impedance loads and be able to work in high temperature environments continuously for long periods of time.

To help assess the performance of power amplifiers, various specifications and standards are given, some of which leave the intended user in a state of confusion rather than enlightenment!

Terminology

When deciding on an amplifier's suitability for use in a given circumstance, the power output is usually viewed first. In general, amplifier projects state r.m.s. power output into a particular load impedance, e.g. 100W r.m.s. into 8Ω, and 150W r.m.s. into 4Ω, etc.

How were these figures derived and what do they mean? Unfortunately, for the average user there are many ways of measuring the rated output and power capability of an amplifier. Generally, the most meaningful are based on Average Continuous Wave Power using a constant sine wave signal and resistive loads. The formula then used for average power W_{av} is $W_{av} = VI$ where V is the measured r.m.s. voltage across the load and I is the r.m.s. current through the load, or $W_{av} = V^2/R$ and $W_{av} = I^2R$. R is used to represent the load resistance in ohms.

R.m.s. (Root Mean Square) terms are used to distinguish between Peak power and Average power levels, and an r.m.s. power figure will be a lot less than the instantaneous or peak power level. The actual peak voltage (see Figure 2) across the load is $\sqrt{2}$ of the r.m.s. voltage, so peak power levels are expressed as

$$W_{PK} = \frac{V^2\sqrt{2}}{R}$$

where V is the r.m.s. voltage, and in theory W_{PK} is twice that of average power.

To go a stage further, r.m.s. power values require the instantaneous (peak) power to be squared, integrated and the resultant found from its square root, all very complicated, but simplified:

$W_{RMS} = W_{av} \times \sqrt{1.5}$ or $W_{RMS} \simeq 1.225 W_{av}$

Let us now use these formula in real terms to see how they work.

Average Power

$W_{av} = V^2/R$ where V is the r.m.s. voltage across R. The r.m.s. voltage is derived from the sine wave voltage in Figure 2. If we take for this example the

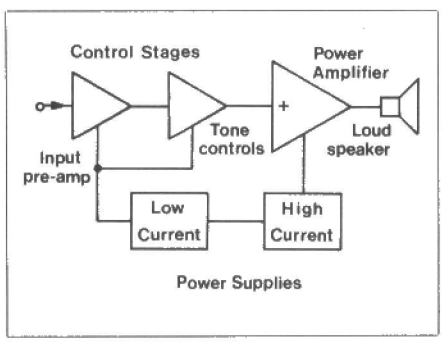


Figure 1. Driving a loudspeaker.

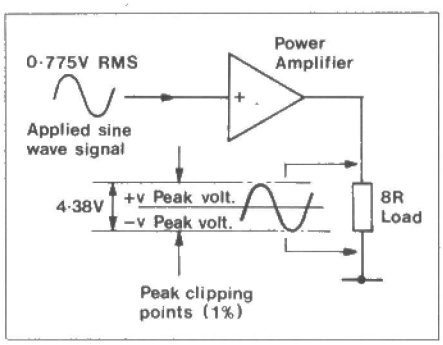


Figure 2. Peak voltage.

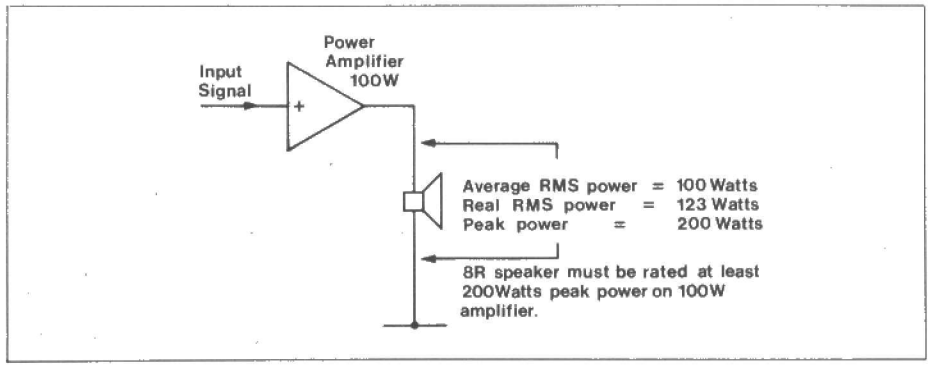

Input Signal
Power Amplifier 100W
Average RMS power = 100 Watts
Real RMS power = 123 Watts
Peak power = 200 Watts
8R speaker must be rated at least 200Watts peak power on 100W amplifier.

Figure 3. 1.225 times the rated output in Real power or double in Peak power.

peak to peak voltage across R to be 4.38V (measured on an oscilloscope) then the instantaneous peak voltage will be half the peak to peak voltage for the +V peak and half again for the −V peak or 4.38/2V. Each peak voltage = 2.19V.

$$\text{As } V_{RMS} = \frac{V_{PK}}{\sqrt{2}}$$

$$\text{the r.m.s. voltage} = \frac{2.19}{1.414} \text{ or } 1.55V_{RMS}$$

The average power is now calculated from:

$$W_{av} = \frac{(V_{RMS})^2}{R}$$

$$\text{Therefore: } W_{av} = \frac{1.55^2}{8} = 0.3 \text{ Watts r.m.s.}$$

Peak Power

The peak voltage of a sine wave is $\sqrt{2}$ of the r.m.s. voltage, therefore peak

$$\text{power} = \frac{(V_{RMS})^2\sqrt{2}^2}{R}$$

$$\text{i.e. } W_{PK} = \frac{1.55^2 \times 2}{8} = 0.6 \text{ Watts,}$$

$$\text{or } W_{RMS} \times \sqrt{2}^2 = 0.3 \times 2 = 0.6 \text{ Watts.}$$

The peak power is twice that of the r.m.s. power.

Real r.m.s. Power

$$W_{RMS} = \frac{(V_{RMS})^2 \times 2}{2R} \times \sqrt{1.5}$$

The formula expresses Real r.m.s. power which is greater than average power as follows:

$$W_{RMS} = \frac{1.55^2 \times 2}{16} \times \sqrt{1.5}$$

$$\simeq 0.37 \text{ Watts,}$$

$$\text{or } W_{RMS} \times \sqrt{1.5} = 0.3 \times \sqrt{1.5} = 0.37 \text{ Watts.}$$

To sum up then, an amplifier specified in average or just r.m.s. power will deliver 1.225 times the rated output in real r.m.s. terms or twice the rated output in peak power. These points are important when choosing loudspeakers to be used with amplifiers rated in average (r.m.s.) Watts, especially at high power levels (see Figure 3).

Music Power

Another commonly used term is Music Power, which refers to a maximum obtained output from a sine wave signal for short durations, referred to a particular distortion figure. Of importance in this specification, is the power supply used with the amplifier.

Assume that the theoretical power supply, shown in Figure 4 generates +50V DC across C1, and −50V DC across C2, and from which the amplifier is drawing very little current (no input signal). The perfect supply would be able to maintain 100V DC under all load conditions, but this supply is not regulated, therefore as the amplifier current increases, with a signal present at the input, both C1 and C2 will discharge and the 50V supply rails will drop in voltage for a short period. This temporary loss limits the maximum signal that can be delivered before the sine wave starts to clip and distortion is incurred. C1 and C2 will not discharge immediately as they are usually of high capacitance (10,000μF and more being typical), so for a short period, the amplifier can deliver a larger signal than is possible under *continuous* signal conditions. The actual increase or Music Power is approximately 30% more than the average r.m.s. power, but this percentage is dependent on supply and load regulation. In a stereo amplifier, it is usually possible for one channel only to develop a higher power output than it could with both channels driven.

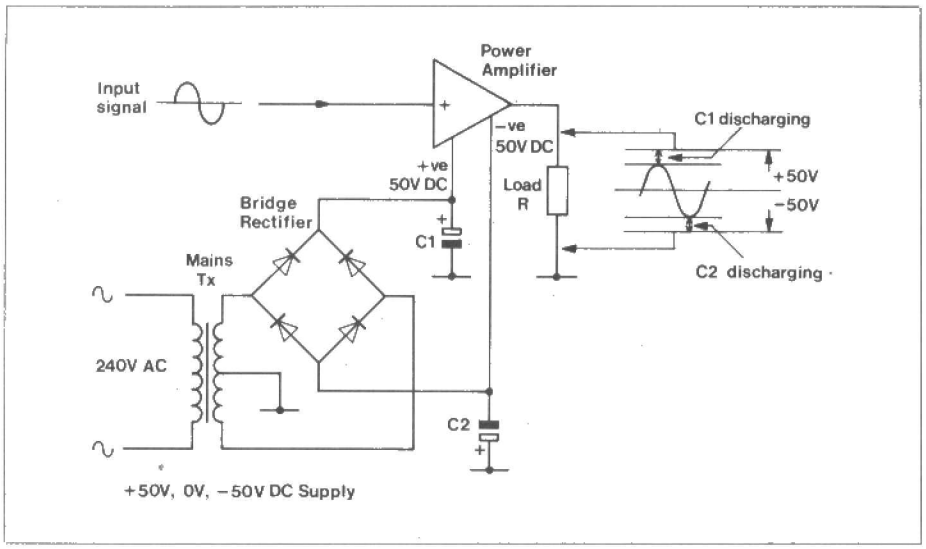


Figure 4. Theoretical power supply.

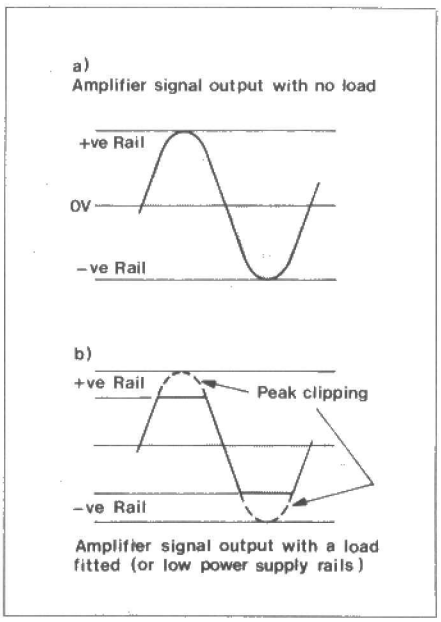


Figure 5. Clipping.

Specifications, therefore, often state power output with one channel only being driven and both channels driven simultaneously. In this case, the single channel rating can be stated in Music Power, whereas the stereo (both channels) rating stated in Average Continuous Wave Power or r.m.s., because the PSU has to supply both amps.

Clipping

This term is often used in all types of analogue circuit descriptions and refers to the point on a sine wave where the maximum supply is exceeded (see Figure 5). Waveform (a) represents an amplifier signal present at the amplifier output when no load is applied. Waveform (b) shows the same waveform when either the supply rails are low or the load applied is excessive. It could be described as the point where the amplifier's output stage literally runs out of supply voltage 'headroom'. This effect is common when several speakers are paralleled together and their combined impedance is lower than the amplifier can handle. Clipping flattens the tips of the waveform, turning a sinewave into a close representation of a square wave in extreme cases! Square waves are exceedingly rich in harmonics and the audible effect of clipping is similar to the classic rock guitarist's fuzz or overdrive sound. Not desirable in Hi-Fi systems! Exceeding a power amplifier's input signal level rating will also produce clipping and in some circumstances, this will damage power devices within the amplifier itself or even the speakers.

To review on the parameters discussed so far, we are looking for:

(a) The average and peak power output into a specific load.
(b) The maximum input signal level applied to achieve the rated output.
(c) PSU required for driving one or two amps before clipping occurs.

Distortion

Various types of waveform distortion are always present in any amplification system. Terms such as Total Harmonic Distortion (THD), Inter Modulation Distortion (IMD), and Transient Distortion (TD) are often used to describe the amplifier's capability in handling these parameters.

Generally with power amps, the THD figure offered should be better than 0.1% at a frequency of 1kHz (1000 cycles per second) and is measured with a distortion factor test set. This method of measurement encompasses all harmonics within a particular bandwidth and generated noise, which give an overall distortion factor. At low signal levels, the noise figure is significant in the factor, but a more accurate figure can be derived from the formula:

Distortion factor =

$$\sqrt{\frac{(\text{Noise}^2 + \text{r.m.s. sum}^2)}{\text{r.m.s. Signal.}}}$$

Noise is the noise voltage, r.m.s. sum of harmonic voltages and total r.m.s. signal voltage.

Real THD is found from squaring the sum of all harmonic voltages, subtracting the square of noise voltage divided by r.m.s. signal voltage and finding the square root of the result.

When two or more different sine waves are present in an amplifier, their linearity can be affected. This is more likely to occur with complex music waveforms and the effect known as Inter Modulation Distortion, IMD, is often more useful than THD in specifications, although the inter-modulation components are difficult to appraise without specialised wave analysers and extensive use of poly-nomials. One severe type of IMD is crossover distortion, shown in Figure 6. This effect is more common in amplifiers having Push-Pull output stages and is caused by a mismatch in the transfer of the signal between the positive and negative polarity halves of the stage (various types of output stages used in power amplifiers will be viewed later on in the series). Transient inter-modulation distortion becomes a problem when pre-amp or control stages have a much higher frequency response than the Open Loop response of the power amplifier. Input stages within the amplifier become overloaded and this condition exceeds the Open Loop Rise Time causing bursts of signal modulation, very unpleasnatn to the listener.

Adding to the list of parameters then, we now have:

(d) Distortion figures, usually THD better than 0.01% in modern amplifiers.

Slewing Rate

Very common now in high frequency audio power amplifiers is the term Slew

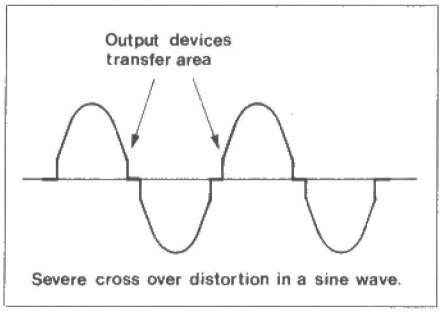


Figure 6. Crossover distortion.

Rate. Although the audio bandwidth is generally specified from 20Hz to 20kHz, the full Power Bandwidth usually extends far above the audible range and the slew rate refers to the maximum frequency at which an amplifier delivers a voltage relating to its full power output rating.

The formula is Slew Rate =

$$\frac{V_{PK} \times 2\pi \text{ Freq}_{max}}{10^6} \text{(per } \mu\text{s)}$$

This can be used where V_{PK} equals the peak voltage across the load, Freq_{max} is the maximum frequency at which full power output is maintained (cycles per second) and 10^6 introduces the time factor in microseconds (μs) because Slew Rate actually relates to a rate of change in time. The constant PI(π) is also used for the maximum slope of a sine wave at 2π radian.

EXAMPLE:

Using this formula in a typical example, assume we have a 100W rated power amplifier with full power bandwidth of 40kHz driving into an 8Ω load.

First, calculate the peak voltage, V_{PK} from $W = \frac{(V_{RMS})^2}{R}$

By transposition $V_{RMS} = \sqrt{W \times R}$

And therefore $V_{PK} = \sqrt{WR} \times \sqrt{2}$

$V_{PK} = \sqrt{100 \times 8} \times 1.414$

$V_{PK} \simeq 40V$

The Slew Rate, SR =

$$\frac{40V \times 2\pi \times 40{,}000Hz}{10^6} \text{(per } \mu\text{s)}$$

When this is calculated, it produces a Slew Rate figure of 10 Volts per microsecond or 10V/μs.

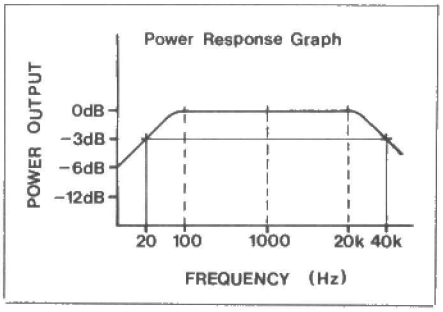


Figure 7. Power response.

Frequency Response

This well used term relates to the minimum and maximum operational area of the power amplifier as shown in Figure 7. The graph shows an essentially flat response from 100Hz to 20kHz with gradual slope off (−3dB) at 20Hz to 40kHz. Modern technology has produced semiconductors capable of operating well into high frequency areas and it becomes necessary to reduce the amplifiers response to prevent radio breakthrough problems. Of more interest is the Full Power Bandwidth or Power Response specification which is defined as the range between low and high frequencies where the power output level is 3dB down (−3dB) from the rated power (measured at 1kHz) although the 3dB figure is not usually given.

Our parameter list now extends to:

(e) The slewing rate, indicating HF performance.

(f) Frequency response, full power bandwidth measured from 1kHz.

Further parameters will be highlighted later on in the series, but enough has been discussed so far to enable us to look at a practical amplifier project based on the TDA2030 power IC.

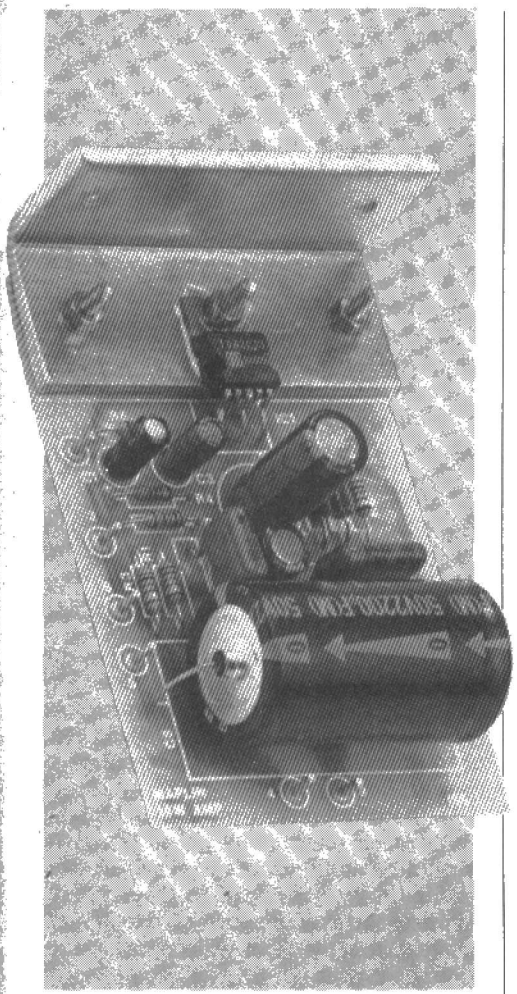
15W Power Amplifier

SPECIFICATION

Power Supply Voltage	-	36V DC Absolute Max
Supply Current, No Signal/Load	-	30 to 40mA Min
Full Load	-	1.5A Max
Short Circuit Current	-	1A Average
Short Circuit Duration	-	Continuous, Shut Down at 110°C
Power Output Rating	-	14W r.m.s. into 8Ω continuous, 20W r.m.s. into 4Ω continuous
THD	-	0.2% at 1kHz
Power Response	-	20Hz to 100kHz
Slewing Rate	-	9.5V/μs
Input Sensitivity	-	250mV for rated output

This specification is taken from Maplin's standard 15W Amplifier Kit (YQ43W) which is quite simple to build and of relatively low cost. For non-constructors, the module is also available in ready-built form (YQ37S), and is fully tested and working. Figure 8 shows the circuit diagram of the amplifier and Figure 9 details a simple power supply, for which a PCB only (no kit) is available (YQ38R).

Specifications such as this are taken under laboratory test conditions using analysing equipment and supply stan-



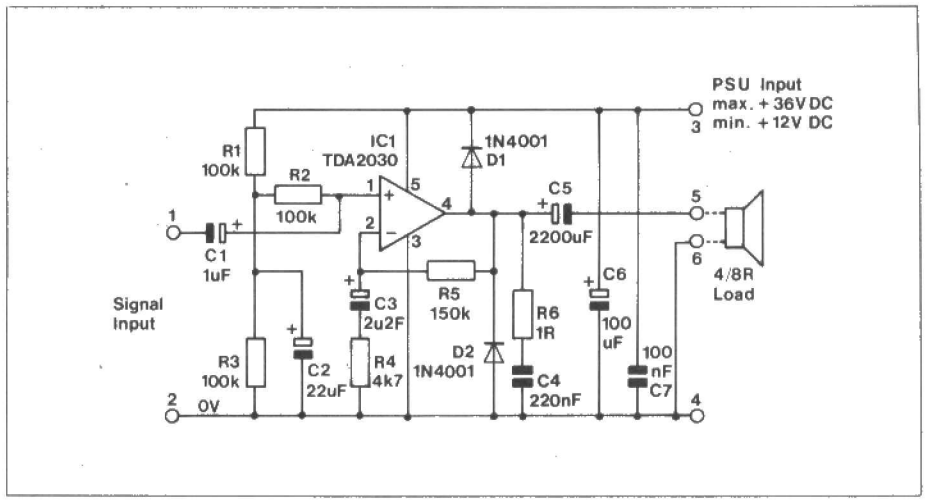


Figure 8. 15W Power Amplifier circuit.

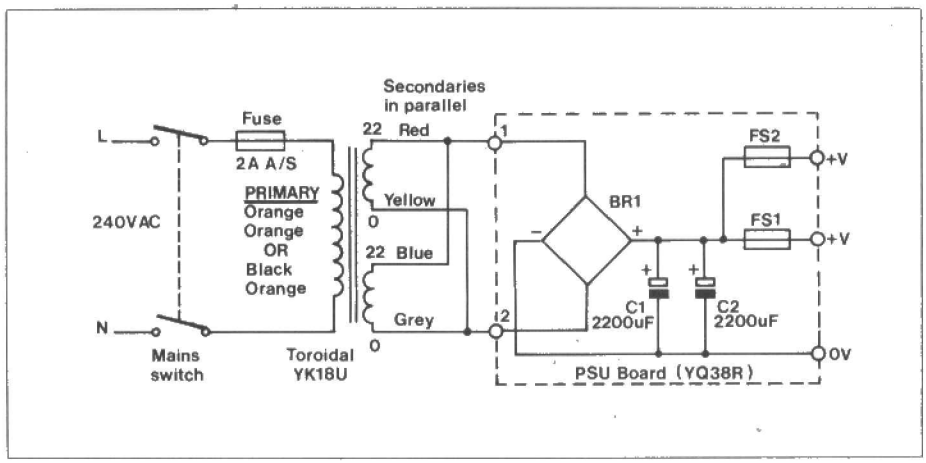


Figure 9. Power supply for 15W amplifier.

dards. These conditions are hardly relevant to the numerous possible situations that an amplifier is likely to encounter in every day use, but a general idea of performance can be taken from this. For instance, the PSU shown in Figure 9 is not regulated and will not maintain exactly 35.5V DC under varying load situations! Maximum output figures have been taken just below the peak clipping point (0.2% distortion) so PSU voltage levels are critical for the specification. A more realistic power output would be around 10W r.m.s. into 8Ω loads and 15W r.m.s. into 4Ω loads, allowing for low mains supply, poor transformer regulation and variables in component tolerances. Remember that a full wave rectified and smoothed supply only will sag as greater current demands are placed upon it, hence the voltage level will drop. Larger value capacitors can sometimes be fitted for greater current availability but power on surge currents may damage the rectifier diodes and pop off fuses.

Stereo

If two amplifiers are to be run from the given PSU then current availability will have to be shared by both amplifiers. As the PSU is only capable of supplying one amplifier at 20 Watts, it could not possibly supply two amps for 20W output each. In this circumstance the maximum possible output from each amplifier would be around 12W into 4Ω loads each (total 24 watts). Speakers used on each amp must be able to handle peak power levels in excess of 25 Watts as peak power is twice that of average r.m.s. figures.

To continue with the specification, the 36V DC absolute max figure means just that! A small spike on the mains could be superimposed onto the positive supply rail, thus exceeding the max rating and damaging IC1. Suitable suppression would have to be fitted in noisy environments to help prevent this from happening.

The no signal or load parameter is taken after shorting the input pin to ground to prevent stray pick-up, while the output is left open. A quiescent current of 30 to 40mA is normal on this amplifier and indicates the minimum current requirement. Full load indicates the maximum current likely to be required by the amplifier and therefore assists in deciding on which type of PSU should be used. Short circuit current is lower because the IC will current limit its output when driving heavy loads or into a short circuit to 0V. As a further safeguard, the TDA2030 also shuts down operations when case temperatures on power devices must be kept low by attaching suitable heatsinks and this is often overlooked or under-provided. As a rough guide, consider the heat output from a 15W or 40W domestic lamp, which is quite considerable. More on heatsinks later on in the series. The rest of the specification has been covered previously except for Input Sensitivity. In this case, 250mV is requried for full output ratings and exceeding this figure will cause clipping at the output and severe distortion. 250mV is again an r.m.s. figure and viewed on an oscilloscope would be measured as 0.707V peak to peak.

As can be appreciated, understanding amplifiers and their specifications can present a formidible task for the uninitiated. The tendency is to look for the largest power output figure at the lowest financial outlay and trust the specification to tell you exactly what you want to know, regardless of how it is interpreted!

This series will cover more specification parameters, different classes of power amp output stages, heatsinking, screening and earth loop problems. More power amplifier modules from Maplin's range will be reviewed and a new giant 500/1000W system introduced using high power MOSFET devices. Methods of bridging amplifiers will also be discussed and it is intended that many of the regular technical problems encountered by constructors will be catered for along the way.

Tungsten Lamp Surge Controller

by R.T. Irish

The Principles of Phase Control

There are frequent practical requirements to control the electrical power delivered to a load - an electric lamp, an electric motor or an electrical heater for example. If the supply is DC from a battery say, this control can simply be achieved by connecting a resistor, R, in series with the device, as shown in Figure 1.

If the supply voltage is V, it may easily be shown that the power delivered to the load, R_L is:

$$P = \frac{R_L}{(R + R_L)^2} \times V^2 \text{ Watts}$$

Two extreme cases show the power controlling action, firstly when R = 0 and the power becomes V^2/R_L (as expected) and secondly, when R = infinity and the power in R_L is zero. Unfortunately, the simple consideration of these two extremes masks the situation when R has some practical value between zero and infinity. The current in the control resistor R causes power to be dissipated in it and the Power Efficiency of the system is less than 100%. In fact:

Power Efficiency =

$$\frac{\text{Power to Load}}{\text{Power Input}} = \frac{R_L}{R + R_L} \times 100\%$$

This situation, in which the power efficiency is significantly less than 100%, is sometimes improved by switching V on and off with an appropriate duty cycle.

Similar considerations apply when using alternating voltages to supply a load, but in this case, effective control of the duty cycle is much easier to implement using semiconductor switching devices, such as the silicon-controlled-rectifier (s.c.r.) or the triac. The most usual method of power control

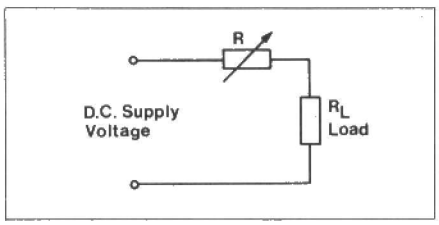


Figure 1. The DC Control Problem.

is phase control.

Figure 2a shows an alternating supply voltage feeding a load resistor (R_L) via the switch S. When S is open, the power dissipated in R_L is zero and when S is closed the power dissipated in R_L is V^2/R_L, where V is the r.m.s. value of the AC supply voltage.

Imagine it was possible to open the switch at the start of a supply cycle, point A in Figure 2b, and to close it again at point B. Between B and C the switch is kept closed. For this first half cycle, the load will only be supplied with power for half the time, and the average power over this half-cycle would be one half of the power which would be supplied if the whole cycle were applied to R_L. Opening the switch at C and closing it again from

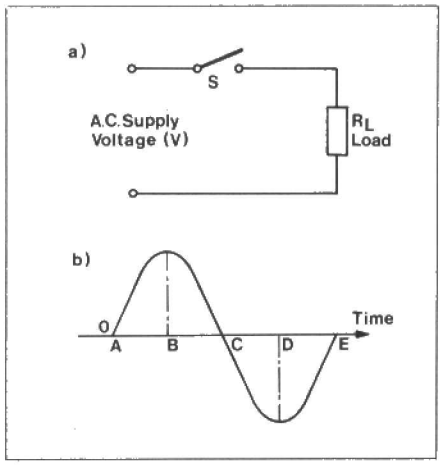


Figure 2. Principles of Phase Control.

D to E would produce the same result for the second half cycle. Thus the average power to the load has been halved but the efficiency is 100% (assuming a perfect switch).

Extending this principle further, there is no reason why points B and D should be at the cycle's peaks as shown in Figure 2b, moving B to the left will increase the average power fed to R_L and moving it to the right will decrease the average power.

This method of controlling the average power is known as phase control. If the r.m.s. supply voltage is V_{RMS} then it may be easily shown that the r.m.s. voltage applied to the load is:

$$V_{RMS} \sqrt{\frac{2(\pi - \propto) + \sin 2\propto}{2\pi}}$$

Where $\propto$ is the angle at which switch S is closed (i.e. B or D). A graph showing how the r.m.s. load voltage varies as $\propto$ is varied between zero and 180 degrees (0 and π radians) is shown in Figure 3.

As the r.m.s. voltage varies with the angle $\propto$, so also does the power. The ratio of the actual power dissipated in R_L to the maximum power (V^2/R_L) is also shown in Figure 3 for completeness.

The switching, at the appropriate firing angle, $\propto$, is achieved by the use of a unijunction transistor (u.j.t.) which applies a pulse to the gate of the s.c.r. or triac at the correct time. Switch-off is automatic when the supply voltage deactivates the s.c.r., or triac, as it goes through zero at the end of the half-cycle.

Lamp Control and Surge Suppression

It is well known that the resistance of the filament of a tungsten lamp is very much lower when cold, than when at its operating temperature. This effect causes a high current to flow when the lamp is first switched on and the thermal

shock produced is frequently enough to break, prematurely, the brittle filament.

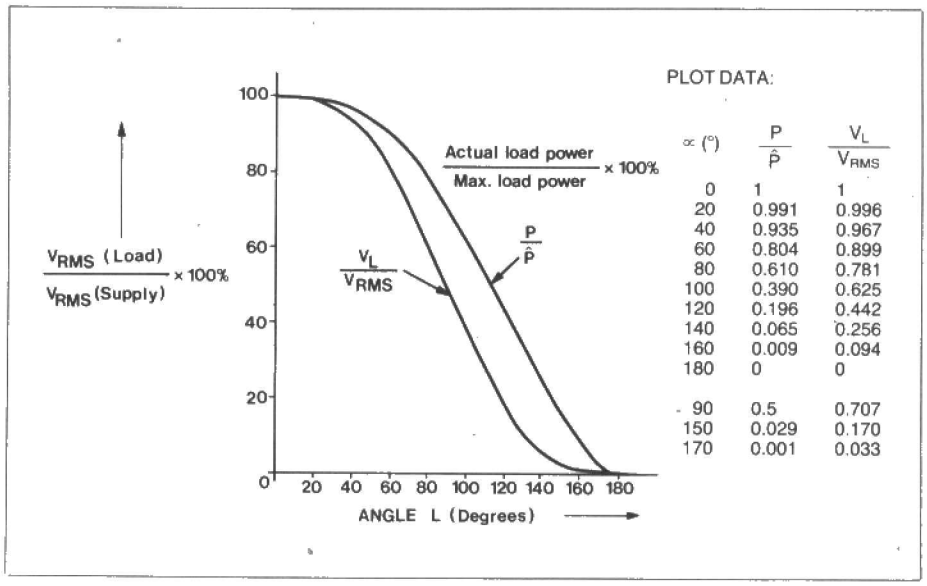
The recent installation of a new domestic lighting system, using some very expensive bulbs, prompted the author to design a controllable, 'soft-start' circuit which overcomes this difficulty. In this circuit the supply voltage is gradually applied to the lamp, effectively eliminating the switch-on current surge and thus increasing the life of the lamp.

Circuit Description

The circuit, shown in Figure 4, consists of a mains fed bridge rectifier which feeds the zener diode ZD1 through resistor R1. As the mains sinusoidal voltage rises from zero, the voltage across ZD1 also rises until the zener's breakdown voltage is reached. For the continuing further rise of the mains voltage, the voltage across ZD1 remains constant at 18V, producing the waveform shown in Figure 5.

The remainder of the circuit may be understood by considering the simplified circuit shown in Figure 6. At the beginning of each mains cycle C1 is discharged and the rising, and eventually stabilised voltage across ZD1, causes C2 to charge through R4. When the breakdown voltage of the unijunction is reached (about 12V in this circuit) it fires and discharges C2 through the primary of the pulse transformer. This, in turn, fires the triac from the transformer's secondary and hence current flows through the lamp for the remainder of the cycle. If R4 is made adjustable, then the average current through the lamp may be controlled - since this controls the charging rate of C2. In the complete circuit, R4 is a preset resistor which controls the effective voltage applied to the lamp at switch-on.

The rate at which the voltage, applied to the lamp, builds up may be understood by referring to Figure 7. Here I1 is a LM334Z constant current source, controlled by the preset R3. Initially, C1 is discharged and the triac firing is controlled by R4 and C2, as has already been described. Under these conditions all of the current through I1 flows into C1 through D1, D2 is reverse biased, isolating R4, C2 and the unijunction. As time continues, so the charge on C1 gradually builds up and provides an ever-increasing starting voltage for the charging of C2, reducing the firing angle of the unijunction and increasing the brightness of the lamp. The current through I1, controlled by R3, therefore determines the rate at which the lamp increases its brightness. The use of a constant current device in this position is preferred to the use of a simple resistor because of the better 'feel' of the current control by R3. A resistor in this position does not seem to behave quite as expected because of the exponential nature of the charging of C1 which would then occur.



∝ (°)	$\frac{P}{\hat{P}}$	$\frac{V_L}{V_{RMS}}$
0	1	1
20	0.991	0.996
40	0.935	0.967
60	0.804	0.899
80	0.610	0.781
100	0.390	0.625
120	0.196	0.442
140	0.065	0.256
160	0.009	0.094
180	0	0
90	0.5	0.707
150	0.029	0.170
170	0.001	0.033

Figure 3. Effective Load Voltage and Power verses Firing Angle.

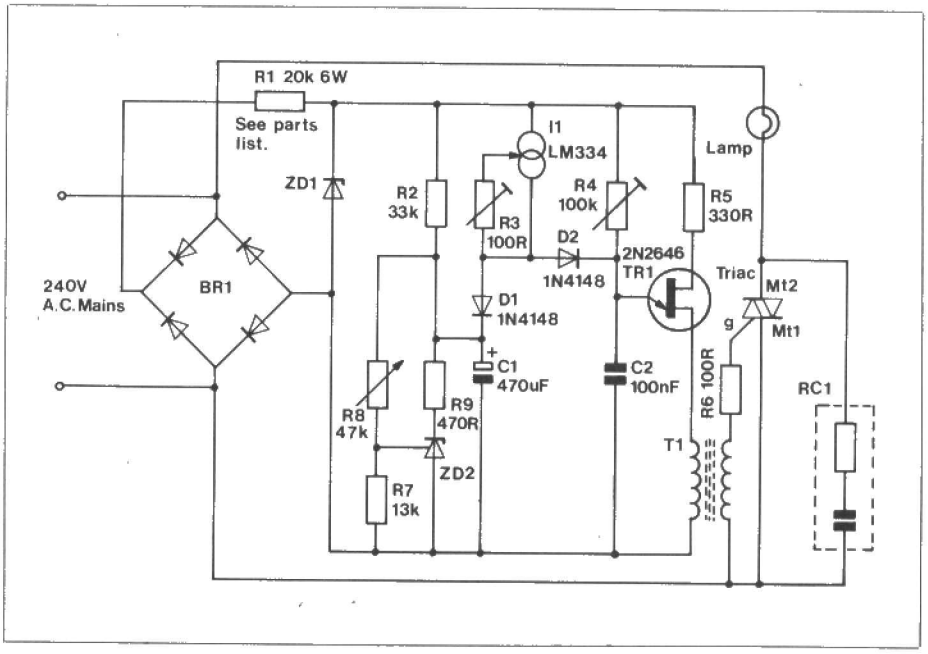


Figure 4. Control/Delay Circuit.

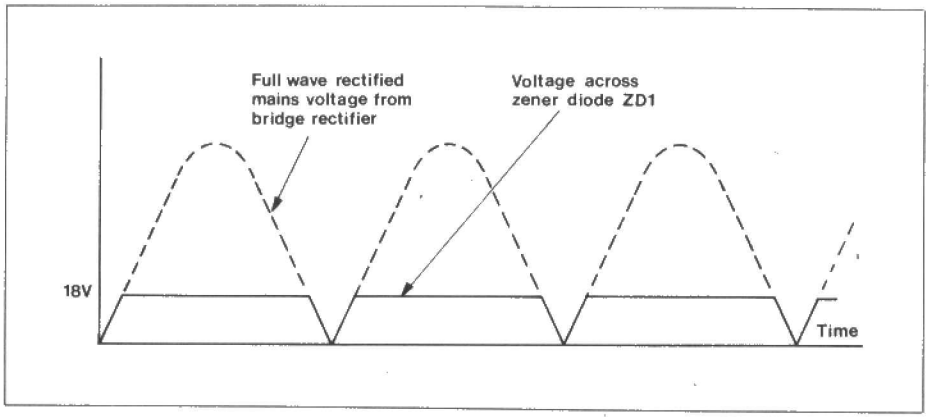


Figure 5. Voltage across ZD1.

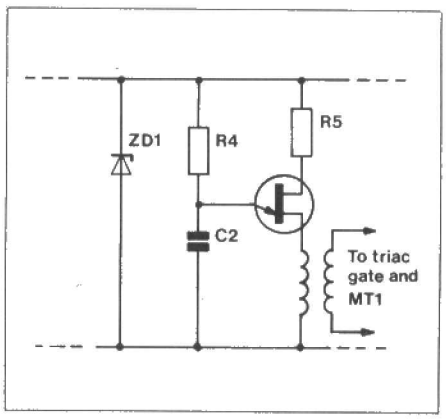


Figure 6. Basic Triac Firing Circuit.

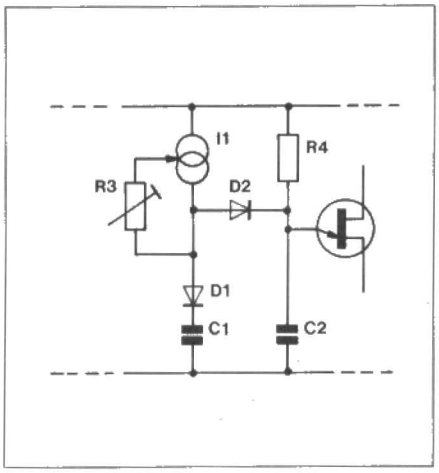


Figure 7. Delayed Triac Firing Circuit.

The circuit shown in Figure 7 would continue increasing the voltage applied to the lamp until it was virtually equal to the whole mains voltage. In order to control the final brilliance of the lamp, the zener diode ZD2 plus R7 and R8 are connected across C1. These components limit the maximum voltage to which C1 charges with the current provided by I1.

It is useful to summarise the actions of the relevant components in the timing operations involved:

R4 and C2 govern the effective voltage applied to the lamp at the instant of 'switch-on'.

R3 and C2 govern the small delay between the mains voltage rise and its application to the lamp when fully lit (ZD2 inoperative).

R3 and C1 determine the time taken for the voltage supplies to the lamp to rise to its maximum value.

ZD2 and R7, R8 limit the maximum voltage across C1 and thus control the lamp's eventual brightness.

The 470R resistor, R9, is included to limit the discharge current from C1 if R8 should be suddenly reduced.

Building the Circuit

It is essential that the circuit should be completely safe and therefore that no part of it should be readily accessible. This implies:

a) that any pcb you design or other circuit construction, should be mounted on insulating pillars, holding it well clear of any surrounding metalwork.

b) that this device should be fused so that in the event of a mishap the whole circuit will be made safe.

Additionally, it must be borne in mind that some ventilation of the case will be required to help dissipate the heat generated in R1 and in the triac.

The triac itself must be mounted on a suitable heatsink using an insulating mica mounting kit (details are to be found in the Constructors Guide leaflet) but clamping it to a metal panel will provide adequate heatsinking for most applications. With some effort, it is possible to obtain similar triacs whose heat dissipating surfaces are electrically isolated and these may be attached directly to a metal container making a compact unit. In the event that such an insulated triac is not available, care must be taken to include fuses which would blow in the event of the insulating washer breaking down.

Setting up the Circuit

Connect a short-circuit across C1 and adjust R4 until the lamp is just on. This sets the brightness of the lamp at switch-on (some constructors may prefer to adjust this control until the lamp is just extinguished).

Remove the short-circuit from C1 and set R8 to its maximum value, rendering ZD2 inoperative. Switch on the system and observe the lamp running up to its full brightness. The speed of this run may be adjusted by R3. Note however, that too small a value of R3 will produce a 'snap-on' effect. A run up time of about 5 seconds for most lamps is suggested. R8 is the brightness control which may include a mains switch if this is considered desirable.

Observations

The applications for this most versatile circuit encompass all those for simple power control, where the slow-start components may be omitted. Alternatively, if a simple slow-start is required, where maximum power is always eventually needed, then R7, R8 and R9 may be omitted. Such applications include high power lighting for stages, football stadia, etc., when the switch-on current surge is frequently high enough to damage the feed cables and mains switches.

Using this circuit, the author has had no lamp failures in the last eight months and there is no doubt that the system will eventually pay for itself - both in the cost of new lamps and in the frayed tempers whenever lamps fail at the inevitably inconvenient moment.

PARTS LIST

	Description	Stock Codes
BR1	Bridge Rect. 400V 1.5A W04	(QL40T)
R1	20k, 6W (two 10k 3W in series)	
R2	33k	(M33K)
R3	100R Cermet Preset	(WR38R)
R4	100k Cermet Preset	(WR44X)
R5	330R	(M330R)
R6	100R	(M100R)
R7	13k	(M13K)
R8	47k Pot not switched	(FW04E)
	switched	(FW44X)
R9	470R	(M470R)
C1	470μF, 25V	(FF16S)
C2	0.1μF	(YR75S)
T1	Pulse Transformer	(HX81C)
ZD1	Zener Diode, 18V, 400mW	(QH20W)
ZD2	TL430 Adjustable Zener Diode	(YY77J)
TR1	2N2646	(QR14Q)
D1,2	1N4148	(QL80B)
Triac	- to suit lamp current e.g.	
	C206D for lamps up to 3A	(WQ24B)
	Heatsink	(FL58N)
I1	LM334Z Adjustable Current Source	(WQ32K)
RC1	Contact Suppressor (0.1μF and 100R)	(YR90X)

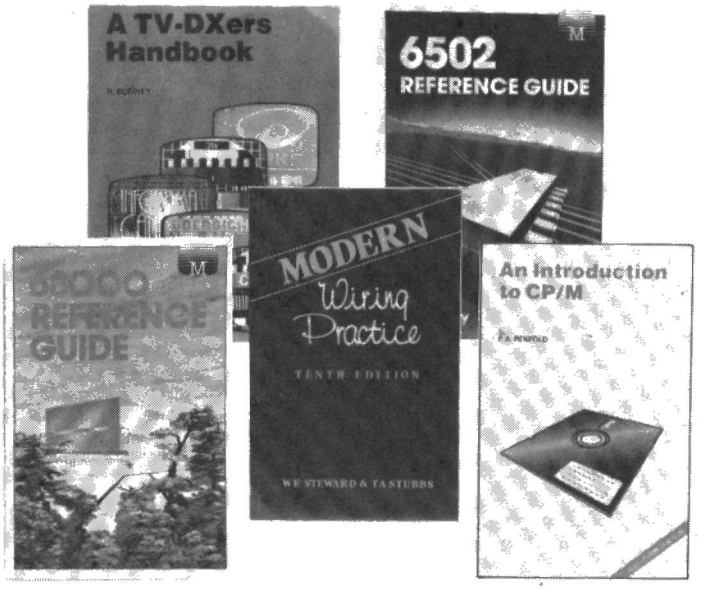
MAPLIN'S TOP TWENTY KITS

THIS MONTH	LAST		DESCRIPTION OF KIT	ORDER CODE	KIT PRICE	DETAILS IN PROJECT BOOK
1.	(1)	⇔	Live Wire Detector	LK63T	£3.50	14 (XA14Q)
2.	(2)	⇔	U/Sonic Car Alarm	LK75S	£16.95	15 (XA15R)
3.	(3)	⇔	100W Mosfet Amplifier	LW51F	£17.95	Best of E&MM
4.	(4)	⇔	Partylite	LW93B	£9.95	Best of E&MM
5.	(6)	⇑	Car Burglar Alarm	LW78K	£7.95	4 (XA04E)
6.	(7)	⇑	Ultrasonic Intruder Detector	LW83E	£11.95	4 (XA04E)
7.	(5)	⇓	8W Amplifier	LW36P	£5.50	Catalogue
8.	(–)	⇑	I/R Prox. Detector	LM13P	£10.95	20 (XA20W)
9.	(10)	⇑	PWM Motor Driver	LK54J	£9.95	12 (XA12N)
10.	(14)	⇑	Logic Probe	LK13P	£13.95	8 (XA08J)
11.	(16)	⇑	15W Amplifier	YQ43W	£6.50	Catalogue
12.	(8)	⇓	Car Battery Monitor	LK42V	£7.50	Best of E&MM
13.	(13)	⇔	Burglar Alarm	LW57M	£59.95	2 (XA02C)
14.	(17)	⇑	Car Digital Tacho	LK79L	£19.95	Best of E&MM
15.	(18)	⇑	Syntom	LW86T	£13.95	Best of E&MM
16.	(12)	⇓	Noise Gate	LK43W	£10.95	Best of E&MM
17.	(19)	⇑	Servo and Driver	LK45Y	£10.95	11 (XA11M)
18.	(–)	⇑	Computadrum	LK52G	£10.95	12 (XA12N)
19.	(9)	⇓	Stepper Motor and Driver	LK76H	£15.95	18 (XH18U)
20.	(–)	⇑	Musical Announcer	LK57M	£13.95	13 (XA13P)

Over 150 other kits also available. All kits supplied with instructions.

The descriptions above are necessarily short. Please ensure you know exactly what the kit is and what it comprises before ordering, by checking the appropriate Project Book mentioned in the list above – see inside back cover for details.

New Books



A TV-DXers Handbook

by R. Bunney

This book examines the possibilities and problems of the long-range reception of TV signals, and the technique of resolving these pictures for the minimum distortion on the TV screen. The satellite TV section has been greatly expanded to encompass further generalised information, since in the coming decade the direct reception of satellite TV signals by the viewer at home may be the next major change in broadcasting techniques. The book also includes many circuits and devices designed by experienced enthusiasts for the specific purpose of long-range TV reception, and illustrating how particular individual problems were overcome.

Order As WP61R (TV DXers Handbook) ***Price £5.95 NV***

68000 Reference Guide

by Richard Woolcock and Anthony Burkitt

The 68000 microprocessor with its 32-bit architecture is at the heart of a rapidly growing range of new, low cost, personal and home computers such as the Sinclair QL, Atari 520ST and Commodore Amiga - machines calculated to make all the 8-bit machines we have just got used to as good as obsolete in a very short space of time. The tremendous speed and versatility of the 68000 is such that many are regarding it as *the* processor of the future.

For owners of such 68000 equipped machines (and even possibly for an experimenter's 68000 'breadboard'), this book examines 68000 architecture and machine code in detail, with due emphasis given to the instruction set and addressing modes, and the complicated business of interrupts. One section is devoted to a detailed breakdown of 68000 hardware, and another to useful programming, illustrating the 68000's powerful abilities, and showing some really quite simple basic constructs.

Finally four appendices are provided for quick reference of instruction codes, 68008 and 68020 variants, and a table of binary representations of all object codes. Invaluable for the experienced programmer and newcomers alike.

200 x 141mm, 204 pages, illustrated.

Order As WP63T (68000 Ref Guide) ***Price £12.95 NV***

6502 Reference Guide

by Alan Tully

A must for any 6502 based micro-computer owners who want to try some serious machine code programming is this comprehensive machine code reference guide. The book covers 6502 architecture, CPU timing principles, and both a brief summary as well as full detailed descriptions of all the instructions, and how the processor status flags operate. Also provided are some helpful hints and tips for assembly language programming, and instruction and ASCII code tables. Applicable to the Apple II, II+ and IIe; BBC model B, Acorn Electron, Atari 400 and 800, Commodore 64, VIC 20 and PET.

200 x 140mm, 240 pages, illustrated.

Order As WP62S (6502 Ref Guide) ***Price £10.95 NV***

Modern Wiring Practice

by W.E. Stewart and T.A. Stubbs

A completely revised edition of *Modern Wiring Practice* taking into account the requirements of the 15th edition of the IEE Regulations for Electrical Installations, and amendments to those regulations published up to January 1986. Includes new procedures developed since the ninth edition was published.

Both the design of electrical installation systems and the practical work itself are covered. The authors have many years' experience of electrical installation, and therefore pass on their experience and recommendations for high standards of design and workmanship in all aspects of electrical installation. An absolute must for all electrical engineers, apprentices, contractors and architects etc., and an invaluable reference for DIY wiring at home.

205 x 138mm, 294 pages, illustrated.

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An Introduction to CP/M

by R.A. Penfold

For newcomers to computing, and even for those with some form of programming experience, it can be difficult at first to understand an operating system such as CP/M.

The primary function of any operating system is to enable the various parts that make up a complete computer to work together in an orderly and co-ordinated way, and thus enable the computer to perform its tasks efficiently and without crashing.

It is a popular misconception that CP/M is mainly a disk operating system, that it is solely concerned with the control and operation of disk drives, but this is not the case. CP/M can control a full range of peripherals in addition to program control.

It is not necessary to understand how CP/M works in order to be able to use it, but some detailed knowledge, as this book provides, can be of immense help in order to fully exploit the potential of the system.

Order As WP60Q (Intro to CP/M) ***Price £2.95 NV***

CORRIGENDA

Vol. 4 No. 13

Musical Announcer: Some problems may be experienced by constructors where the special synthesiser chip IC1 has been 'latching up' or falsely re-triggering. Modifications to prevent this are as follows - remove C1 (0.1μF) completely. Replace R1 (10k) for Min Res 2k2. Replace R4 (68R) for Min Res 33R. Replace C2 (0.1μF) for tantalum electrolytic of 1μF, mount +lead towards IC1 pin 28.

Vol. 5 No. 20

Satellite Decoder, Sync Tone Board: In the parts list, the stock code for the vertical sub-min. presets RV1 and RV2 should be WR73Q, not WR70Q.

AMENDMENTS TO 1987 CATALOGUE

SLOPING FRONT CASE XY60Q (Page 78). Please note that this case does not have ventilation holes as described in the 1987 catalogue.

4K7 ENCLOSED PRESET UH15R (Page 293). In the description following the stock code this item is referred to as being horizontal mounting, whereas it is in fact vertical mounting.

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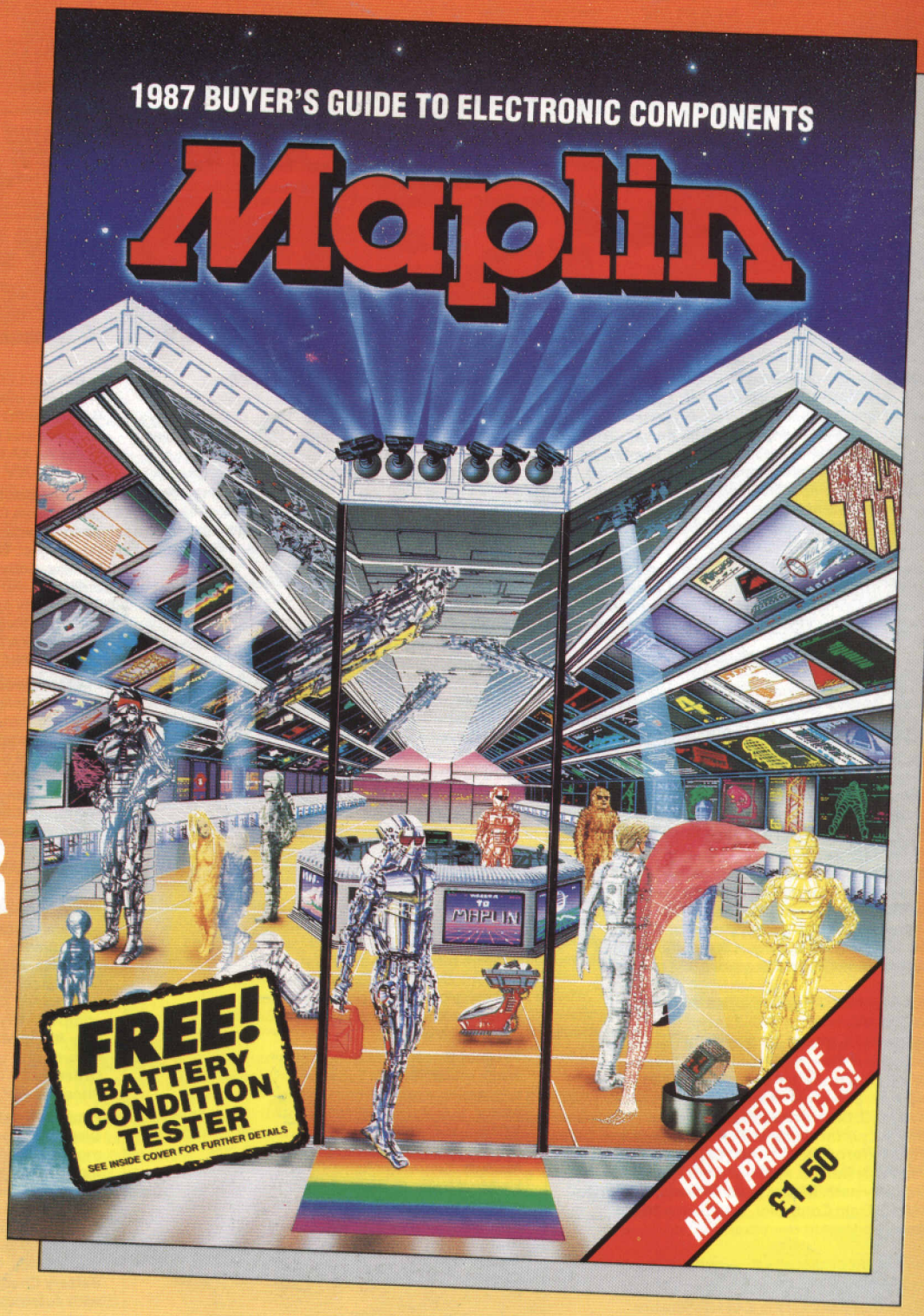

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